



## Circumferential-wave phase velocities for empty, fluid-immersed spherical metal shells

Überall, Herbert; Claude Ahyi, A.; Raju, P. K.; Bjørnø, Irina; Jensen, Leif Bjørnø

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## Session 2aAAa

**Architectural Acoustics: Halls for Music Performance . . . Another Two Decades of Experience 1982–2002  
(Poster Session)**

Christopher A. Storch, Cochair

*Artec Consultants, Inc., 114 West 26th Street, New York, New York 10001*

Timothy J. Foulkes, Cochair

*Cavananaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776*

Ian B. Hoffman, Cochair

*The Talaske Group, Inc., 105 North Oak Park Avenue, Oak Park, Illinois 60301***Contributed Papers**

All posters will be on display from 9:00 a.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon. To allow for extended viewing time, posters will remain on display until 5:00 p.m.

**2aAAa1. Severance Hall, Cleveland, OH.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa6. Kultur- und Kongresszentrum Luzern, Switzerland.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa2. Squires Recital Salon, Blacksburg, VA.** Noral D. Stewart (Stewart Acoust. Consultants, P.O. Box 30461, Raleigh, NC 27622, noral@ix.netcom.com)

**2aAAa7. New Jersey Performing Arts Center, Newark, NJ.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa3. Joan and Irving Harris Concert Hall.** Elizabeth A. Cohen (Cohen Acoust., Inc., 132 S. Lucerne Blvd., Los Angeles, CA 90004, akustik@mediaone.net) and David Schwind (Charles M. Salter Assoc., Inc., San Francisco, CA 94914)

**2aAAa8. The Morton H. Meyerson Symphony Center, Dallas, TX.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa4. Sibelius Hall, Lahti, Finland.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa9. Symphony Hall, Birmingham, United Kingdom.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa5. LG Arts Center, SangNam Hall, Seoul, Republic of Korea.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa10. The Winspear Centre for Music.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa11. The Chan Centre for the Performing Arts, Vancouver, British Columbia.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa12. Auditorium de Dijon, Dijon, France.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa13. Sala Sao Paulo, Sao Paulo, Brazil.** Christopher Storch (Artec Consultants, Inc., 114 W. 26th St., 9th Fl., New York, NY 10001-6812)

**2aAAa14. Hobby Center for the Performing Arts, Houston, TX.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa15. The Church of Jesus Christ of Latter-Day-Saints Assembly Hall, Salt Lake City, UT.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa16. Bass Performance Hall, Ft. Worth, TX.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa17. Oklahoma City Civic Center Hall, Oklahoma City, OK.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa18. Tokyo International Forum, Tokyo, Japan.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa19. John F. Kennedy Center Concert Hall, Washington, DC.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa20. Zankel Recital Hall, at Carnegie Hall, New York City, NY.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa21. River Center for the Performing Arts, Columbus, GA.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa22. Murchison Performing Arts Center, University of North Texas, Denton, TX.** Mark Holden (Jaffe Holden Acoustics, Inc., 114A Washington St., Norwalk, CT 06854, mholden@jhacoustics.com)

**2aAAa23. Paul F. Sharp Concert Hall, University of Oklahoma, Norman, OK.** (Acentech, Inc., Cambridge, MA)

**2aAAa24. Rogers Performing Arts Center, Merrimack College, North Andover, MA.** (Acentech, Inc., Cambridge, MA)

**2aAAa25. Weis Auditorium, Bucknell University, Lewisburg, PA.** (Acentech, Inc., Cambridge, MA)

**2aAAa26. Center for Performing Arts, Univ. of Florida, Gainesville, FL.** (Acentech, Inc., Cambridge, MA)

**2aAAa27. Spivey Hall, Clayton College and State University, Morrow, GA.** (Acentech, Inc., Cambridge, MA 02138)

**2aAAa28. College of New Jersey Recital Hall, Ewing, NJ.** (Acentech, Inc., Cambridge, MA 02138)

**2aAAa29. Auckland Town Hall.** Christopher W. Day, Ewen R. Kitto, and Keith O. Ballagh (Marshall Day Acoust., P.O. Box 5811, Wellesley St., Auckland, New Zealand, auckland@marshallday.co.nz)

**2aAAa30. Bruce Mason Theatre—Takapuna.** Christopher W. Day, Joanne O. Valentine, and Keith O. Ballagh (Marshall Day Acoust., P.O. Box 5811, Auckland, New Zealand)

**2aAAa31. Saginaw Valley State University, Recital Hall, Saginaw, MI.** Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301)

**2aAAa32. Coronado Theatre, Rockford, IL.** Richard H. Talaske and Jonathan P. Laney (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301)

**2aAAa33. Goshen College, Concert Hall and Recital Hall, Goshen, IN.** Gary S. Madaras and Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301)

**2aAAa34. Central Michigan University, Concert Hall, Mt. Pleasant, MI.** Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301)

**2aAAa35. California Center for the Arts, Concert Hall, Escondido, CA.** Richard H. Talaske (The Talaske Group, Inc., 105 N. Oak Park Ave., Oak Park, IL 60301)

**2aAAa36. Meymandi Concert Hall, Raleigh, NC.** Edward Dugger (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa37. Verbrugghen Hall.** Edward McCue (Kirkegaard Assoc., 954 Pearl St., Boulder, CO 80302)

**2aAAa38. Concert Hall at the Clarice Smith Performing Arts Center.** Edward McCue (Kirkegaard Assoc., 954 Pearl St., Boulder, CO 80302)

**2aAAa39. Blumenthal Performing Arts Center, Charlotte, NC (1992).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa40. Auer Concert Hall.** Edward McCue (Kirkegaard Assoc., 954 Pearl St., Boulder, CO 80302)

**2aAAa41. Great Hall of the Washington Pavilion.** Edward McCue (Kirkegaard Assoc., 954 Pearl St., Boulder, CO 80302)

**2aAAa42. Jacoby Symphony Hall, Jacksonville, FL.** Edward Dugger (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa43. Barshinger Center for Musical Arts, Lancaster, PA.** Edward Dugger (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa44. Merrill Auditorium, Portland, ME.** Edward Dugger (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa45. Broward Center for the Performing Arts, Ft. Lauderdale, FL (1991).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa46. Cerritos Center for the Performing Arts, Cerritos, CA (1993).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa47. Concert Hall at the Frank L. Moody Music Building.** R. Lawrence Kirkegaard (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa48. Ordway Music Theatre, St. Paul, MN (1985).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa49. Corbett Auditorium, University of Cincinnati College-Conservatory of Music, Cincinnati, OH (1996).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa50. Orchestra Hall in Chicago, IL.** Dawn Schuette (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa51. Aronoff Center for the Performing Arts.** Dawn Schuette (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa52. Benedict Music Tent at the Aspen Music Festival.** Dawn Schuette (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa53. North Shore Center for the Performing Arts.** Dawn Schuette (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa54. Dewan filharmonik petronas (Petronas Concert Hall), Kuala Lumpur, Malaysia (1998).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa55. Maui Community Arts and Cultural Center, Kahului, HI (1994).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa56. Lagerquist Concert Hall, Pacific Lutheran University, Tacoma, WA (1995).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa57. Davies Symphony Hall, San Francisco, CA.** Martha Larson (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa58. Peace Center for the Performing Arts, Greenville, SC (1990).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa59. Shepherd School of Music, Rice University, Houston, TX (1991).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa60. Heinz Hall, Pittsburgh, PA.** Carl Giegold (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa61. Joseph M. Meyerhoff Symphony Hall, Baltimore.** Carl Giegold (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa62. St. Luke Centre, London.** Carl Giegold (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa63. Barbican Concert Hall, London.** Carl Giegold (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa64. Jordan Hall at the New England Conservatory.** Scott Pfeiffer, Clete Davis, and Larry Kirkegaard (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607)

**2aAAa65. Jemison Concert Hall, University of Alabama, Birmingham, AL (1996).** Joseph W. A. Myers (Kirkegaard Assoc., 801 W. Adams St., 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

**2aAAa66. Casals Hall, Tokyo, Japan.** Toshiko Fukuchi, Keiji Oguchi, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, fukuchi@nagata.co.jp)

**2aAAa67. Sumida Triphony Hall, Tokyo, Japan.** Toshiko Fukuchi, Hideo Nakamura, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, fukuchi@nagata.co.jp)

**2aAAa68. Nara Centennial Hall, Nara, Japan.** Satoru Ikeda, Chiaki Ishiwata, Motoo Komoda, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-ku, Tokyo)

**2aAAa69. The Sogakudo Concert Hall of the Tokyo National University of Fine Arts and Music, Tokyo, Japan.** Tosiko Fukuchi, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10 Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, fukuchi@nagata.co.jp)

**2aAAa70. Tokyo Metropolitan Art Space, Tokyo, Japan.** Satoru Ikeda, Keiji Oguchi, Makoto Ino, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo, Japan)

**2aAAa71. The Toppan Hall, Tokyo, Japan.** Tosiko Fukuchi, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10 Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, fukuchi@nagata.co.jp)

**2aAAa72. Tsuda Hall, Tokyo, Japan.** Satoru Ikeda, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo)

**2aAAa73. Philia Hall, Tokyo, Japan.** Akira Ono, Katsuji Naniwa, Takeshi Yamamoto, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10 Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, ono@nagata.co.jp)

**2aAAa74. Katsushika Symphony Hills, Tokyo, Japan.** Satoru Ikeda, Akira Ono, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo Bunkyo-ku, Tokyo 113-0033, Japan, ikeda@nagata.co.jp)

**2aAAa75. Asahikawa City Taisetsu Crystal Hall, Hokkai-do, Japan.** Akira Ono, Makoto Ino, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo Bunkyo-ku, Tokyo 113-0033, Japan, ono@nagata.co.jp)

**2aAAa76. Kioi Hall, Tokyo, Japan.** Akira Ono, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo Bunkyo-ku, Tokyo 113-0033, Japan, ono@nagata.co.jp)

**2aAAa77. Kumamoto Prefectural Theatre, Kumamoto, Japan.** Satoru Ikeda, Hideo Nakamura, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa, Bldg. 3F, 2-35-10 Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, ikeda@nagata.co.jp)

**2aAAa78. Concert hall ATM in Art Tower Mito, Mito, Japan.** Keiji Oguchi, Hideo Nakamura, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10 Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, oguchi@nagata.co.jp)

**2aAAa79. Okayama Symphony Hall, Okayama, Japan.** Satoru Ikeda, Keiji Oguchi, Hideo Nakamura, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa, Bldg. 3F, 2-35-10 Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, ikeda@nagata.co.jp)

**2aAAa80. The Harmony Hall, Matsumoto, Japan.** Toshiko Fukuchi, Keiji Oguchi, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa, Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, fukuchi@nagata.co.jp)

**2aAAa81. Fukushima Concert Hall, Fukushima, Japan.** Yasuhisa Toyota, Keiji Oguchi, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, toyota@nagata.co.jp)

**2aAAa82. Nagaoka Lyric Hall, Nagaoka, Japan.** Yasuhisa Toyota, Chiaki Ishiwata, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, toyota@nagata.co.jp)

**2aAAa83. Sapporo Concert Hall “Kitara,” Sapporo, Japan.** Yasuhisa Toyota, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, [toyota@nagata.co.jp](mailto:toyota@nagata.co.jp))

**2aAAa84. Suntory Hall, Tokyo, Japan.** Yasuhisa Toyota, Akira Ono, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, [toyota@nagata.co.jp](mailto:toyota@nagata.co.jp))

**2aAAa85. Akiyoshidai International Art Village, Yamaguchi, Japan.** Keiji Oguchi, Makoto Inoh, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, [oguchi@nagata.co.jp](mailto:oguchi@nagata.co.jp))

**2aAAa86. Ishikawa Concert Hall, Kanazawa, Japan.** Keiji Oguchi, Satoru Ikeda, Masaya Uchida, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, [oguchi@nagata.co.jp](mailto:oguchi@nagata.co.jp))

**2aAAa87. Kyoto Concert Hall, Kyoto, Japan.** Yasuhisa Toyota, Keiji Oguchi, Chiaki Ishiwata, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, [oguchi@nagata.co.jp](mailto:oguchi@nagata.co.jp))

**2aAAa88. Queensland Conservatorium of Music Theater, Brisbane, Australia.** Keiji Oguchi, Yasuhisa Toyota, Katsuji Naniwa, and Minoru Nagata (Nagata Acoustics, Inc., Hongo Segawa Bldg. 3F, 2-35-10, Hongo, Bunkyo-Ku, Tokyo 113-0033, Japan, [toyota@nagata.co.jp](mailto:toyota@nagata.co.jp))

**2aAAa89. Mitaka City Arts Center, Tokyo, Japan.** Takayuki Hidaka (Takenaka R&D Institute, 1-5-1, Otsuka Inzai Chiba 270-1395, Japan, [hidaka.takayuki@takenaka.co.jp](mailto:hidaka.takayuki@takenaka.co.jp))

**2aAAa90. New National Theater, Tokyo, Japan.** Takayuki Hidaka (Takenaka R&D Institute, 1-5-1, Otsuka Inzai Chiba 270-1395, Japan, [hidaka.takayuki@takenaka.co.jp](mailto:hidaka.takayuki@takenaka.co.jp))

**2aAAa91. Hamarikyu Asahi Hall, Tokyo, Japan.** Takayuki Hidaka (Takenaka R&D Institute, 1-5-1, Otsuka Inzai Chiba 270-1395, Japan, [hidaka.takayuki@takenaka.co.jp](mailto:hidaka.takayuki@takenaka.co.jp))

**2aAAa92. Tokyo Opera City (TOC) Concert Hall, Japan.** Takayuki Hidaka (Takenaka R&D Institute, 1-5-1, Otsuka Inzai Chiba 270-1395, Japan, [hidaka.takayuki@takenaka.co.jp](mailto:hidaka.takayuki@takenaka.co.jp))

**2aAAa93. Rocky Hill High School Auditorium—Rocky Hill, CT.** Bennett M. Brooks (Brooks Acoust. Corp., 27 Hartford Turnpike, Vernon, CT 06066)

**2aAAa94. Crowell Concert Hall, Wesleyan University Center for the Arts, Middletown, CT.** Bennett M. Brooks (Brooks Acoust. Corp., 27 Hartford Turnpike, Vernon, CT 06066)

**2aAAa95. Kavli Theater at Thousand Oaks Civic Arts Plaza, Thousand Oaks, CA.** Ron McKay and Mark Rothermel (McKay Conant Brook, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, [mrothermel@mcbin.com](mailto:mrothermel@mcbin.com))

**2aAAa96. Royce Hall at University of California Los Angeles, Westwood, CA.** Ron McKay and Mark Rothermel (McKay Conant Brook, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, [mrothermel@mcbin.com](mailto:mrothermel@mcbin.com))

**2aAAa97. Popejoy Hall at University of New Mexico, Albuquerque, NM.** Ron McKay and Mark Rothermel (5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, [mrothermel@mcbin.com](mailto:mrothermel@mcbin.com))

**2aAAa98. Kaul Auditorium, Reed College, Portland, OR.** Russ Altermatt (Altermatt Assoc., Inc., 522 SW Fifth Ave., Ste. 1200, Portland, OR 97204, [raltermatt@altermatt.com](mailto:raltermatt@altermatt.com))

**2aAAa99. Acoustical renovation of The Orpheum Theatre, Vancouver, Canada.** John O’Keefe (Aercooustics Eng. Ltd., 50 Ronson Dr., Toronto M9W 1B3, Canada), Gilbert Soloudre (Carleton Univ., Ottawa K1S 5B6, Canada), and John Bradley (Inst. for Res. in Construction, Natl. Res. Council of Canada, Ottawa K1A 0R6, Canada)



**2aAAa100. The chamber music/lecture hall for the Royal Library in Copenhagen, Denmark.** Anders Christian Gade (Acoust. Technol., Oersted DTU, Tech. Univ. of Denmark, Bldg. 352, DK-2800 Lyngby, Denmark)

**2aAAa102. Evans Concert Hall, Connecticut College.** Timothy Foulkes (Cavanaugh Tocci Assoc., 327F Boston Post Rd., Sudbury, MA 01776)

**2aAAa101. Radiohuset, Studio 1, Copenhagen.** Jens H. Rindel, Anders C. Gade (Oersted-DTU, Acoust. Technol., DTU Bldg. 352, DK-2800 Lyngby, Denmark), and Niels V. Jordan (Jordan Akustik, Herslevvej 19, Gevninge, DK-4000 Roskilde, Denmark)

**2aAAa103. Performance Hall, Providence, RI.** Lincoln B. Berry (Cavanaugh Tocci Assoc., 327 F. Boston Post Rd., Sudbury, MA 01776, berry@cavtocchi.com)

TUESDAY MORNING, 5 JUNE 2001

RED LACQUER ROOM, 9:00 A.M. TO 12:00 NOON

### Session 2aAAb

## Architectural Acoustics, Engineering Acoustics and Noise: Acoustical Test Facilities (Poster Session)

Charles T. Moritz, Chair

*Blachford, Inc., 1400 Nuclear Drive, West Chicago, Illinois 60185*

### Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

**2aAAb1. Johns Manville Acoustical Laboratory.** Brandon D. Tinianov and Francis J. Babineau (Johns Manville Tech. Ctr., 10100 W. Ute Ave., Littleton, CO 80127)

Due to their long history in acoustics materials, the Johns Manville Corporation has maintained an acoustical laboratory in constant operation since the 1930s. The laboratory complex is accredited under NVLAP for a broad range of ASTM and ISO test methods. For classic architectural acoustics evaluation, the lab has a reverberant test suite with a 142-m<sup>3</sup> source chamber and a 316-m<sup>3</sup> receive chamber. The chambers' common opening allows for 2.8 by 4.3-m samples and is qualified from 100–10 000 Hz. Analysis of complex, three-dimensional systems conducted in a 176-m<sup>3</sup> hemi-anechoic chamber qualified to 160 Hz. Both intensity and pressure measurements are possible via a 5 degree-of-freedom robotic positioning system. Research of open office acoustics and office furniture is performed in a 111- (to the grid, 142 total) m<sup>3</sup> chamber designed to accept a suspended ceiling and vary its reverberation time from <0.3 to 2.3 s. In addition to these facilities, several benchtop and analytical resources are available.

**2aAAb2. Acoustic test facilities of the Underwater Sound Reference Division.** A. Lee Van Buren, Robert M. Drake, and Kirk E. Jenne (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841-1708)

The acoustic test facilities of the Underwater Sound Reference Division (USRD) of the Naval Undersea Warfare Center will be described. The USRD serves as the U.S. standardizing activity in the area of under-

water acoustic measurements, as the National Institute of Standards and Technology does in other areas. It is the Navy's primary activity for underwater acoustic calibration, test, evaluation, and reference measurements on transducers and related devices and materials. In this role the USRD maintains specialized measurement facilities with the capacity to simulate real-world ocean environments. It also maintains transducer standards to help ensure the accuracy of measurements made both at USRD and elsewhere. This presentation will describe the Acoustic Pressure Tank Facility, the Low-Frequency Facility (with its three test vessels), the Acoustic Open Tank Facility, the Long-Line Hydrophone Calibrator, the Conical Shock Tube Facility, the Leesburg Facility, and the Transducer Standards Loan Program.

**2aAAb3. Acoustic measurement facilities at the Applied Research Laboratory.** David L. Bradley (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030)

Acoustic measurement facilities, ranging from a water tunnel that is also listed as an ASME Historic Landmark to a new ultrasonic measurement system, are illustrated and described with a brief explanation as to size, measurement equipment available, and frequency ranges that are typically employed. The facilities include a high-pressure tank for hydrophone tests "at depth," the ultrasonics tank, an anechoic tank for hydrophone calibration, the water tunnel, an air acoustic anechoic room and a large, multipurpose underwater measurement facility.

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**2aAAb4. Energistics Laboratory facility.** Victoria Cerami (Cerami & Assoc., Inc., 317 Madison Ave., Ste. 220, New York, NY 10017) and Jack Stegall (Energistics Lab., Houston, TX 77257)

Energistics Laboratory in Houston, Texas is a leading laboratory for the testing of HVAC equipment. For over 15 years, this facility has ensured the highest standards in leading-edge HVAC technology and architectural testing capabilities. Testing capabilities include both industry standard rating procedures, and mock-up testing to simulate field conditions. The laboratory is open to developers, owners, architects, engineers, general contractors, manufacturers and others who require independent component testing and evaluation.

**2aAAb5. Designing acoustic facilities for education.** Daniel R. Raichel (Raichel Technol. Group and CUNY Grad. Ctr., 2727 Moore Ln., Fort Collins, CO 80526)

An acoustic laboratory in a university generally serves two functions: education and research. In planning, budgetary considerations constitute a decisive factor. A low budget can result in the laboratory being relegated to fairly simple measurements but some degree of ingenuity can widen the scope of experimentation, as has been proven in Take Five demonstrations at ASA meetings. A more elaborate facility would enable measurements in the ultrasonic region in addition to those in the audio range. The cost of maintenance and eventual upgrading should be figured into the planning and design of any facility. Necessary instrumentation include means of measuring sound pressure levels and intensities, executing spectral analyses (preferably in real time), generating and synthesizing signals, performing fairly standard measurements for sound absorption coefficients; and it would be desirable that specific enclosure conditions be simulated (viz., through the use of anechoic and reverberation chambers). Research programs generally require more sophisticated instrumentation, but fortunately fairly recent advances in microelectronics permit integration of acoustic monitoring functions into personal computers at considerably lower cost than would be the case if individual measurement instruments were purchased separately.

**2aAAb6. NASA Glenn Research Center Acoustical Testing Laboratory.** Beth A. Cooper (NASA, John H. Glenn Res. Ctr. at Lewis Field, M.S. 86-10, 21000 Brookpark Rd., Cleveland, OH 44135, beth.a.cooper@grc.nasa.gov)

The Acoustical Testing Laboratory (ATL) at the National Aeronautics and Space Administration (NASA) John H. Glenn Research Center in Cleveland, Ohio supports the low-noise design of science experiment payloads developed for the International Space Station (ISS). The ATL consists of a 100-Hz vibration-isolated hemianechoic test chamber with 21-ft by 17-ft by 17-ft (h) interior working dimensions and removable floor wedges that allow the facility to be configured as either a hemianechoic or fully anechoic chamber. A separate, sound-isolated control room with outside dimensions of 23-ft by 11-ft by 12-ft (h) doubles as a noise control enclosure when testing articles that require remote connections to high-noise support equipment and services. These characteristics, along with very low design background sound levels, enable the acquisition of accurate and repeatable acoustical measurements on test articles, up to a full ISS rack in size, that produce very little noise. The ATL provides a comprehensive array of acoustical testing services, including sound power level in accordance with precision and engineering grade standards. These capabilities employ a PC-based acoustical data acquisition system that has been customized to perform simultaneous acquisition and real-time analysis of multiple channels of acoustical signals.

**2aAAb7. USG Corp. Research and Technology Center's Acoustical Testing Facility.** Stephen W. Payne, Jr. (USG Corp. Res. and Technol. Ctr., 700 N. Hwy. 45, Libertyville, IL 60048)

The Acoustical Testing Facility at the USG Corp. Research and Technical Center was commissioned in 1996. The Facility contains structurally isolated test chambers housed within an exterior shell. The Facility includes chambers for performing sound transmission loss tests per ASTM E90, sound absorption tests per ASTM C423, ceiling attenuation tests per ASTM E1414 and interzone attenuation tests per ASTM E1111. In addition to testing per ASTM standards, tests can also be performed to ISO Standards. The chambers are designed to provide maximum isolation from external noise. The chamber walls, roof and floor are built of a minimum of 8-in. (200-mm) solid concrete or filled concrete block. In addition the reverberation chamber is mounted on an independent spring system. The Facility is accredited under the National Voluntary Laboratory Accreditation Program (NVLAP) and under the Underwriters Laboratories Client Data Test Program. The main function of this Facility is to provide data for new product and systems development and to provide testing under the UL Acoustical labeling program.

**2aAAb8. Riverbank Acoustical Laboratories.** David L. Moyer (1512 S. Batavia Ave., Geneva, IL 60134, dmoyer@iitri.org)

Riverbank Acoustical Laboratories (RAL) is located in Geneva, Illinois. Constructed in 1918, RAL became the world's first independent laboratory dedicated to the science of architectural acoustics. Riverbank owner, philanthropist Colonel George Fabyan, built the laboratory for its designer, professor Wallace Clement Sabine. The scientific efforts of this renowned Harvard physicist earned him two distinguished titles, acknowledged today as the father of the science of architectural acoustics and the first "modern day" acoustical consultant. The unit of absorption is a Sabin. When the professor died in 1919, his distant cousin, Dr. Paul Earls Sabine, became RAL's director. Retired in 1947, Dr. Sabine turned the operations over to the IIT Research Institute. RAL provides a wide range of acoustical testing services for over 600 clients. The majority of RAL tests are industrial related. RAL provides clients with an accredited, independent testing facility, not associated with any manufacturer or product. Part of RAL's outstanding credibility reflects upon their policies of maintaining strict confidence and proprietary rights for all clients. RAL is accredited by NVLAP, The City of Los Angeles, and others. Also of interest to many clients is RAL's museum dedicated to the Science of Architectural Acoustics. The ASA officially sanctioned the museum in 1989.

**2aAAb9. Qualification of WEAL Reverberation Chamber.** Gary E. Mange (Western Electro-Acoust. Lab., 1711 16th St., Santa Monica, CA 90404)

Western Electro-Acoustic Laboratory (WEAL) has recently qualified a new reverberation chamber. Qualification measurements were performed for absorption testing (ASTM C423) and sound power testing (ANSI S12.31 and S12.32). Results are shown at various stages of the qualification.

**2aAAb10. Georgia Institute of Technology—Integrated Acoustics Laboratory.** Kenneth A. Cunefare and Van Biesel (G. W. Woodruff School of Mech. Eng., The Georgia Inst. of Technol., Atlanta, GA 30332-0405)

The Integrated Acoustics Laboratory of Georgia Tech's School of Mechanical Engineering is a state-of-the-art facility dedicated to acoustic and vibration research. The current centerpiece of the lab is an anechoic chamber with interior dimensions of 17 ft × 17 ft × 12 ft. Laboratory instrumentation includes a 32-channel HP VXI data acquisition system, full LMS CADA-X data acquisition and analysis package, HP-VEE, a two-channel Siglab analyzer, Polytec PSV-200 scanning laser Doppler vibro-

meter, and a host of microphones, accelerometers, shakers, and load cells. The laboratories also feature a brake dynamometer. Two finite-element modeling stations are equipped with MSC/NASTRAN and COMET/Acoustics software for the complete modeling of vibration and acoustics. The laboratory will be expanded this year to include a reverberation room, which will be attached to the existing anechoic chamber by a hatch for transmission loss testing. The expansion will also add a semi-anechoic chamber to the lab.

**2aAAb11. Noise and structural dynamics test facilities at Boeing.** Wendell Miller, Donald Boston, and Charlie Pickrel (The Boeing Co., Seattle, WA 98124)

The noise and structural dynamics laboratories at Boeing provide a wide range of test and measurement services to the Boeing Company. Test data from these laboratories support all phases of the product life cycle across a diverse line of products and applications. The noise laboratory facilities include a low-speed free-jet acoustic wind tunnel, several anechoic and reverberation test chambers, a critical listening facility, and a materials test center. These facilities are supported with a network of data systems for in-lab testing and a variety of transportable data systems for field- and airplane-based testing. Structural dynamics laboratory facilities include large strongbacks and structural floors for component vibration testing, sonic fatigue test facilities, and vibration test facilities. These facilities are supported by a network of dedicated data systems for a wide

range of modal, shock, vibration, and fatigue testing. Field tests are supported by a wide range of portable data systems and instrumentation trailers capable of large channel count measurements. This work will provide an overview of the test facilities and measurement capabilities of these laboratories.

**2aAAb12. The new Blachford acoustics laboratory.** Charles Moritz (Blachford, Inc., 1400 Nuclear Dr. W., Chicago, IL 60185)

In 1968, HL Blachford, Inc. constructed their first acoustics laboratory next to their Troy, Michigan plant. This laboratory consisted of a 216-m<sup>3</sup> reverberation room with an adjacent large hemianechoic room approximately 12 m long, 5 m wide, and 4.5 m high. After several ownership changes, a plant that used to belong to HL Blachford, Inc. was purchased in 1995 by HL Blachford Ltd. of Canada, and became the new headquarters of a new company called Blachford, Inc. It quickly became obvious that a new laboratory was required for vehicle testing and acoustical material development. The design of a new facility began in 1997 and was based on the old laboratory; however, the new design incorporates design changes to increase the capacity, capabilities, and quality of the facility. The new laboratory is currently under construction and features a larger hemi-anechoic room, approximately 15 m long, 9 m wide, and 6 m high, and a nonadjacent 200-m<sup>3</sup> reverberation room with several horizontal and vertical test sections for vehicle component and material testing.

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TUESDAY MORNING, 5 JUNE 2001

PDR 9, 8:00 TO 11:50 A.M.

## Session 2aAB

### Animal Bioacoustics and Signal Processing in Acoustics: Role of Signal Processing in Understanding Echolocation

James A. Simmons, Chair

*Department of Neuroscience, Brown University, Box 1953, Providence, Rhode Island 02912*

**Chair's Introduction—8:00**

#### Invited Papers

8:05

**2aAB1. Neuromorphic system for real-time biosonar signal processing.** R. Timothy Edwards (Johns Hopkins Univ., APL 4-234, Laurel, MD 20723-6099, tim@stravinsky.jhuapl.edu) and Gert Cauwenberghs (Johns Hopkins Univ., Baltimore, MD 21218-2686, gert@jhu.edu)

Real-time classification from active sonar echolocation requires tremendous computational power, yet is performed effortlessly by bats and dolphins. Special-purpose, yet programmable and reconfigurable hardware tailored to the parallel distributed nature of the computation performed in biology, is presented. The system contains a front-end cochlear processor implemented as an analog filterbank, and a pattern recognition engine classifying the front-end time-frequency features. The reconfigurable front-end supports up to 34 bandpass channels spanning the 10 to 100 kHz range typical of bat and dolphin sonar. A support vector machine (SVM) classifies the features. In an alternative realization, an integrate-and-fire neural network uses the zero-crossings of the front-end bandpass channel outputs. The internal variables may be represented in numerous ways by reprogramming the hardware, giving the system great flexibility in representing features to match the methods used by various biosonar algorithms. Experimental results demonstrate the capabilities of the processor, classifying the CSS (Panama City) linear frequency modulation (LFM2) data in real-time. Implementation issues for alternative representations of biosonar signals are also discussed. [Work supported by ONR N00014-01-1-0315.]

8:25

**2aAB2. Signal processing by echolocating dolphins.** Whitlow W. L. Au (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734)

Many different types of target discrimination experiments have been performed with echolocating dolphins in order to understand and model their echolocation capabilities. However, not many experiments have been performed that measured differential performance as a function of the magnitude of a single target variable, thus making it very difficult to extract limits of various auditory discrimination capabilities. Nevertheless, sufficient information exists that could provide some insight on important echo cues utilized

by dolphins and what the signal processing may be. Furthermore, other types of experiments emulating some dolphin experiments have also been conducted. Stretched versions of echoes from target used in dolphin experiments have been used in human listening experiments to determine salient cues present in the echoes. Neural network experiments have also been conducted using echoes from the same target used in the dolphin experiments. The most pertinent of these experiments involving dolphins, humans, and neural networks will be discussed in order to gain insight into important target cues and the manner in which these cues might be processed by echolocating dolphins. The discussion will be conducted in light of the results of a recent experiment in which the auditory filter shape of a dolphin for different frequencies was determined.

8:45

**2aAB3. What kinds of auditory mechanisms explain biosonar perception by FM bats?** James A. Simmons, Mark I. Sanderson, and Kyler M. Eastman (Dept. of Neurosci., Box 1953, Brown Univ., Providence, RI 02912, james\_simmons@brown.edu)

A biological SCAT model of target ranging in FM-bat biosonar integrates (1) acoustic ecology with algorithms identified from (2) peripheral auditory representations and (3) central auditory processing to account for biosonar performance. The bat's auditory spectrograms incorporate a near-ideal time-frequency compromise and can outperform a simple matched filter for detection of natural targets. Neural coding of delay appears to have submicrosecond accuracy at brain-stem, midbrain, and cortical levels, and can explain ranging and azimuth performance for single targets. In contrast, inner-ear integration-time convolution required to match echo reception to the structure of natural targets blurs delay resolution to hundreds of microseconds, and this is retained in the recovery time of brain-stem responses. Analysis of field performance and laboratory tests both show the bat's delay resolution to be  $\sim 2\text{--}5$  ms, however, midbrain responses constitute delay lines that can support tracking of prey and both coarse target ranging and deconvolution for fine resolution down to 6 ms. Cortical coincidence-detecting neurons test all combinations of pulse-echo delay and spectral profile contingencies to display coarse and fine range. Their characteristics predict bandwidth-squared dependence of submicrosecond hyperresolution, which is confirmed in jittered-echo experiments. The physical embodiment of individual targets in neural responses appears to be dynamic rather than by place.

9:05

**2aAB4. Roles of neural oscillation in time domain processing in the bat inferior colliculus.** Alexander V. Galazyuk and Albert S. Feng (Beckman Inst., Univ. of Illinois, Urbana, IL 61801)

Central auditory neurons in echolocating bats exhibit pulse-echo delay-tuned responses. Sullivan (1982) proposed that paradoxical latency shift (PLS), characterized by an increase in response latency to loud sounds, is important for this attribute. At present, the mechanism underlying PLS is unclear. The goal of the present study was to identify the mechanism underlying PLS. The responses of 92 neurons in the inferior colliculus of little brown bats to brief tone pulses at the unit's CF over a wide range of sound levels were studied. Of these, 16 neurons displayed unit-specific periodic oscillatory discharges at high sound levels with a characteristic period of 1.3–6.7 ms. The 27 neurons exhibited unit-specific PLS, with quantal latency shift of 1.2–8.2 ms. In 14 neurons showing PLS, unit's responses before, during and after iontophoretic application of bicuculline were investigated. Application of bicuculline abolished the PLS and transformed it into periodic discharges, suggesting that neural oscillation in combination with ordinary inhibition may be responsible for PLS. To further investigate whether intrinsic neural property was responsible for PLS, unit's responses to sounds having different durations were investigated. The result indicates that both intrinsic and extrinsic mechanisms are likely involved in creation of PLS. [Work supported by NIH R01DC00663.]

9:25

**2aAB5. Temporal estimation by a model big brown bat.** Janine M. Wotton, Michael J. Ferragamo, Timothy M. Sonbuchner (Biol. Dept., Gustavus Adolphus College), and Mark I. Sanderson (Brown Univ.)

The big brown bat, *Eptesicus fuscus*, uses echolocation to locate prey and displays extraordinary acuity in the perception of temporal cues in acoustic signals. Behaviorally the bat can detect changes at submicrosecond levels but individual neurons in the inferior colliculus (IC) and cortex operate with much less precision. Most of these cells are poor temporal markers with response variation on the order of a few milliseconds and some in tens of milliseconds. A temporal estimator was created incorporating the response properties of recorded neurons and behaviorally appropriate limitations on the number of echolocation emissions. The response of the neurons can be characterized as probability density functions in time and frequency. The characteristics of these neurons were used to create large simulated populations of IC and cortical neurons that show the full range of recorded variation. The connections between these two populations were simulated using a self-organizing neural network. If more than one IC neuron is required to trigger the response of each cortical neuron then the model operates with resolution of microseconds. Manipulating the firing threshold of cortical cells and the relative population sizes influences the errors in target estimation.

9:45

**2aAB6. Binaural hearing in dolphins.** Patrick W. B. Moore and Randall L. Brill (Marine Mammal Program, SPAWAR Systems Ctr., 49620 Beluga Rd., San Diego, CA 92152-6505)

Binaural hearing is an advantage of having two ears. Human benefits are evident in a 3-dB threshold difference, the ability to localize sound sources in space, and the ability to isolate primary sounds from corresponding echoes. The binaural capabilities of dolphins are relatively unexplored. Studies show that their localization of pure tones underwater is mediated by the same mechanisms observed in terrestrial mammals. Behavioral evidence from free-field localization studies supports reliance on time and intensity cues. Two studies have examined binaural hearing in dolphins using contact hydrophones to isolate the hearing mechanisms. They provided masking level differences (MLDs) comparable to humans and interaural time and intensity difference thresholds that were better than any recorded for terrestrial mammals. Neurophysiological studies using evoked potentials investigated interaural sensitivity and intensity differences as a function of multiple frequency stimuli presented at various angles around the head of a dolphin. Recent

behavioral studies have mapped the acoustic sensitivity around the head and lower jaw. Those results suggest greater sensitivity forward of the panbone, thought to be the site of best reception, and an asymmetry in sensitivity that may be analogous to that observed in other animals such as the barn owl.

10:05–10:15 Break

10:15

**2aAB7. Biomimetic chirplet transforms for environmentally adaptive sonar: A model of singing humpback whales.** Eduardo MercadoIII (Ctr. for Molecular and Behavioral Neurosci., Rutgers Univ.–Newark, 197 University Ave., Newark, NJ 07102, mercado@pavlov.rutgers.edu) and L. Neil Frazer (SOEST, Univ. of Hawaii, Honolulu, HI 96822)

Animals that use echoes to identify and localize objects must be able to do so within a variety of acoustically complex environments. The neural processes underlying this ability can be characterized as an adaptive transformation from a one-dimensional space into an  $n$ -dimensional auditory parameter space. Neurophysiological data suggest that in mammals this transformation is highly overcomplete as well as species and experience dependent; such representations are modeled well by chirplet transforms. Chirplet-based representations of sonar signals and their echoes can be used to generate behaviorally relevant maps of acoustic features. These feature maps, which are similar to representational maps in the auditory cortex, can facilitate separation of echoes from predictable noise sources. A model of sound production and reception in humpback whales is presented to illustrate how such a feature map-based sonar might function. Singing humpbacks space themselves apart from and often approach distant, nonsinging whales, demonstrating that they are able to recognize and localize complex acoustic sources at long ranges in shallow water. Singers also modulate their sound sequences to match others they have heard, demonstrating that they are able to accurately and adaptively represent complex environmental sounds. These abilities suggest that humpback whales are well adapted for both passive and active sonar processing.

### Contributed Papers

10:35

**2aAB8. An investigation of the pulses produced by the least shrews (*Cryptotis parva*).** Mersedeh S. Jalili and Jeanette A. Thomas (Lab. of Sensory Biol., Western Illinois Univ., Quad Cities, Moline, IL, mersedehj@aol.com)

Most echolocation studies have focused on bats and dolphins. Because of technological improvements in ultrasonic sensing and recording equipment, there now are cost-effective approaches towards examining “other” groups of mammals for possible echolocation abilities. In this study, we suggest that echolocation is a primitive characteristic, first appearing in insectivores, the ancestor of all other placental mammals. A few other studies and anatomical, behavioral, and physiological attributes suggest that shrews are likely to echolocate. In captivity, least shrews (*Cryptotis parva*) produce series of pulses. We used a board divided into an inner and outer box of equal area to run 8 least shrews through a set of foraging and orientation experiments. Experiments were in the dark and we monitored the circumstances under which shrews produced pulses using a night-vision video camera. An ANABATII bat detector monitored the presence of pulses, a Marantz cassette recorder documented the sounds, and Audio-scope software on a laptop computer provided real time sonogram and oscillogram displays. The number of pulses and trains, the typical waveform, power spectrum, peak frequency, and bandwidth ( $-3$  dB) were examined among the experiments. Results provided strong evidence that least shrews use simple pulses for both orientation and foraging.

10:50

**2aAB9. The influence of flight speed on the ranging performance of bats using frequency modulated echolocation pulses.** Arjan M. Boonman (Dept. of Animal Physiol., Univ. of Tuebingen, Auf der Morgenstelle 28, 72076 Tuebingen, Germany, arjan.boonman@uni-tuebingen.de), Gareth Jones, and Stuart Parsons (School of Biological Sci., Univ. of Bristol, Bristol BS8 1UG, UK)

Many species of bat use ultrasonic frequency-modulated pulses to measure distance to objects. The flight speed of the bat will result in a compression of the echo, and in compression of the time of flight of the signal, both leading to distortion of the perceived range. By comparing Doppler errors incurred with the SCAT-filterbank model and cross-correlation, it was established that the effects of the echo compression are

virtually independent of the two receiver models. A cross-correlation model was used to investigate Doppler tolerance of the ranging estimation and range-Doppler coupling separately. Range-Doppler coupling is considered a potential error for bats since a reference time ( $T_{ref}$ ) does not seem biologically feasible. Without using a reference time, range-Doppler coupling due to echo compression was found to vary between 10 and 30 ms in six bat species. With additional range-Doppler coupling due to the compression of the time of flight (34 ms at 2 m from the target), this leads to perceived target displacements between 4.5 and 6.6 cm at a flight speed of 6 m/s. Range-Doppler coupling can be reduced by linearizing the curvature and by increasing the ratio highest/lowest frequency of the pulse. The study further revealed how pulse duration, bandwidth and harmonics influence Doppler tolerance and range-Doppler coupling.

11:05

**2aAB10. Experimental simulation of binaural object classification by dolphins.** Gerard J. Quentin (Laboratoire d'Imagerie Parametrique, UMR 7623, 15 rue de l'Ecole de Medecine, 75006 Paris, France) and Whitlow W. L. Au (Hawaii Inst. of Marine Biol., Kailua, HI 96734)

We used the experimental setup designed to simulate ultrasound transmission and binaural reception by dolphins at the Marine Mammals Research Program. Three composite piezoelectric transducers, located in the same plane, were placed at the corners of a triangle such that a distance of 14 cm separates the two receivers. This approximates the presumed spacing between the dolphin sound entrances. The two receiving channels approximate those of dolphins (e.g., similar spectral responses, different receiving patterns between the two receptors). Transmitted signals were similar to dolphin clicks and echoes were recorded simultaneously at the two receivers and digitized at 1 MHz using two data acquisition boards housed in a portable computer. We sought to identify different targets in these preliminary experiments: a 3-in. stainless steel spherical shell filled with water and two copper cylindrical shells with outer diameters equal to

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1.63 and 1 in., respectively. The last two samples have the same thickness ( $\frac{1}{16}$  in.) and are filled either with water or with air. The scattered signals are processed via a simple Fourier transform and analyzed using the resonance scattering theory. A preliminary study of the results is presented.

11:20

**2aAB11. Matched field processing of existing binaural dolphinlike signals for MCM.** A. Tolstoy (ATolstoy Sci., 8610 Battailles Court, Annandale, VA 22003, atolstoy@ieee.org) and W. Au (Hawaii Inst. of Marine Biol., Kailua, HI 96734, wau@hawaii.edu)

A new matched field processing (MFP) approach has been applied to existing (but limited) dolphinlike binaural data. The frequencies range from 25 to 200 kHz with the received quasi-monostatic echo lasting less than 4.0 ms. The data discussed consist of backscatter echoes from a mud bottom, a drum buried in the bottom, and a Manta-like object also buried in the bottom. The MFP approach (using NO modeling of any kind) was designed to extract differences between apparently similar signals. Since the available data show overly strong and different returns for all the targets, we first manipulate the data to bring them more into line with expectations. The new data were then processed to yield a target "template" of scattered returns as a function of time and frequency characterizing the target returns. These templates can easily be used to detect and identify scattering from targets on or in the bottom in low S/N situations. [Work supported by ONR.]

11:35

**2aAB12. Consistency of echolocation signals of CF-FM bat measured by a telemetry system.** Hiroshi Riquimaroux, Kenji Yamasumi, Yoshiaki Watanabe (Fac. of Eng., Doshisha Univ., Kyotanabe, Kyoto 610-0321, Japan), and Liang-Kong Lin (Tunghai Univ., Taichung 407, Taiwan)

The purpose of the present study was to examine the daily consistency of the constant frequency component (CF) of CF-FM bats and to test the significance of a telemetry microphone system (telemike) mounted on the bat's head. We examined sonar sound characteristics of leaf-nosed bats (*Hipposideros terasensis*) when bats were at rest and during flight. Experiments were done within a steel-walled chamber. Their emitted pulses were simultaneously recorded by a fixed microphone at the wall and the telemike. The second harmonic of the CF component (CF2) at rest and during flight varied significantly over several months. When at rest, the fundamental frequency of the bat was about 35 kHz. The second harmonic was the most intense. There was little difference between data taken from two different microphones when at rest. However, during flight we found significant differences between data obtained by the microphone at the wall and the telemike. We could see a systematic change in CF2 frequency during flight through the telemike. Our findings may suggest that the bats compensated for the frequency of the CF component of their pulses by detecting Doppler-shifted CF of returning echoes from the target. We will discuss the individuality and consistency of the CF component during flight and at rest. [The present research was supported by Special Research Grants for the Development of Characteristic Education from the Promotion and Mutual Aid Corporation for Private Schools Japan, a grant to RCAST at Doshisha University from the Ministry of Education, Japan and Doshisha University's Research Promotion Fund.]

TUESDAY MORNING, 5 JUNE 2001

MONROE ROOM, 7:30 TO 11:55 A.M.

## Session 2aAOa

### Acoustical Oceanography and Underwater Acoustics: Benchmarking Range Dependent Numerical Models

Kevin B. Smith, Cochair

*Department of Physics, Naval Postgraduate School, Code PH/SK, Monterey, California 93943*

Alexandra I. Tolstoy, Cochair

*ATolstoy Sciences, 8610 Battailles Court, Annandale, Virginia 22003*

Chair's Introduction—7:30

### Invited Papers

7:35

**2aAOa1. Two approximate solutions for the 3-D field in the ASA benchmark wedge.** Grant B. Deane and Michael J. Buckingham (Scripps Inst. of Oceanogr., La Jolla, CA 92093-0238, grant@mpl.ucsd.edu)

One of the current challenges to the underwater acoustics 3-D numerical modeling community is the development of benchmark geometries and solutions. Benchmark solutions to canonical problems provide a reference against which numerical models can be compared, as well as providing insight into regions or frequency regimes where the models fail. The 3-D penetrable wedge is a good candidate for a benchmark geometry because it corresponds to real-world environments (beaches and continental slopes), is tractable to a variety of numerical modelers, and has been studied analytically. Although there is no exact analytical solution available for the penetrable wedge, several approximate solutions have been developed, which may be regarded as secondary benchmarks. The results from 2 of these models for the 3-D extension of the ASA wedge benchmark will be presented.

**2aAOa2. A two-way integral equation approach to a shelf break problem.** David P. Knobles (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029) and Steven A. Stotts (Appl. Res. Labs., Univ. of Texas, Austin, TX 78713)

Benchmark solutions play an important role in the development, testing, and application of numerical models for range-dependent problems in underwater acoustics. To produce benchmark accuracy propagation solutions requires careful attention to both the theoretical formalism and numerical implementation. The focus of this research is a "benchmark" solution for a two-dimensional (2-D) shelf-break problem in ocean acoustics. The method employed to construct a solution is a two-way energy-conserving coupled mode approach in the form of coupled inhomogeneous integral equations. Solutions of the coupled equations are obtained using a powerful approach originally introduced in nuclear theory and also used to solve simple nonseparable problems in underwater acoustics. The basic integral equations are slightly modified to permit a Lanczos expansion to form a solution. The solution of the original set of integral equations is then easily recovered from the solution of the modified equations. Benchmark quality demands that the errors in each step of the numerical calculation be well quantified, so that the resolution in the transmission loss can be determined. Consistency checks are made in the form of using different numerical schemes to compute the adiabatic or free propagator and mode coupling matrix operator.

8:15

**2aAOa3. PE solutions to some internal-wave benchmark problems.** Gordon R. Ebbeson, David J. Thomson (Defence Res. Establishment Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, ebbeson@drea.dnd.ca), and Gary H. Brooke (A & T/Anteon, Victoria, BC V8V 3K3, Canada)

Benchmarking underwater acoustic propagation models is a necessary stage in the evolution of numerical modeling codes. Candidate benchmark problems should challenge the capabilities of existing models in order to promote the development of improved techniques for reliably simulating sound propagation in realistic oceans. In this paper, a parabolic equation (PE) model is applied to the suite of internal-wave test cases that are offered for numerical consideration by the organizers of the Benchmarking Range Dependent Numerical Models session. The PE calculations are carried out using a code that was originally developed for matched-field processing applications [G. H. Brooke *et al.*, "PECan: A Canadian parabolic equation model for underwater sound propagation," J. Comput. Acoust. (2001)]. The capability of PECan to propagate sound accurately through the benchmark environments is examined for both tonal and broadband signals. The discussion will address issues related to reciprocity, interpolation of the internal-wave sound speed structure, and convergence of solutions as a function of Padé order and range and depth step sizes. Where possible, PECan will be validated against other well-known propagation models to assess its accuracy as well as to appraise the suitability of the proposed internal-wave benchmark problems to test the limits of existing propagation codes.

8:35

**2aAOa4. Verified acoustic propagation predictions for a shelf-break case.** Richard B. Evans (Sci. Appl. Int'l. Corp., 21 Montauk Ave., New London, CT 06320) and David B. King (Naval Res. Lab., Stennis Space Center, MS 39529-5004)

A comparison is made between transmission loss computed with stepwise couple modes and the parabolic equation for several shelf-break cases. The close agreement between these two diverse approximate solutions of the wave equation can be interpreted in one of two ways: Either both are producing the correct result or they are making identical errors. The low probability of the second coincidence is interpreted as verification of both results. The difficulties encountered in obtaining verified propagation results are discussed.

### Contributed Papers

8:55

**2aAOa5. Inaccurate results from benchmark accurate ocean acoustic propagation models: It's the implementation, not the model.** Stanley A. Chin-Bing, David B. King, and Robert A. Zingarelli (Naval Res. Lab., Stennis Space Center, MS 39529-5004, chinbing@nrlssc.navy.mil)

Despite numerous publications that document the benchmark accuracy of several ocean acoustic propagation models, some users of these models continue to get inaccurate results. The problem is not with the model, but with the implementation of the model. The most common mistake in implementing range-dependent acoustic models (e.g., parabolic equation models) is the selection of model parameters (e.g., step size, grid spacing, attenuating bottom depth). Another common mistake is lack of sufficient environmental data to the model. The number of implementation errors could be significantly reduced, if the user would abandon the erroneous belief that the model is a "black box" that will always produce the correct answer in a single run. Knowledgeable users methodically vary the input parameters and make repeated runs until model convergence is evident. It is possible to preselect model parameters, and couple the acoustic model with its environmental databases, so that a general user can obtain an accurate result. The U.S. Navy has done this with its Navy Standard Propagation models for nearly a decade. The models and their revisions

are documented, configuration managed, and verified by applications to over 500 test cases. Selected examples will be presented. [Work supported by ONR/NRL.]

9:10

**2aAOa6. Complex ray solutions to the 3-D wedge ASA benchmark problems.** Evan K. Westwood (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713)

In support of the ASA's effort to benchmark shallow-water, range-dependent acoustic propagation models, a complex ray approach is applied to the three-dimensional wedge problems. The ray model [Westwood, J. Acoust. Soc. Am. **92**, 2212–2222 (1992)] is designed specifically for the case of a point source in a 3-D wedge, where the receiver may lie cross slope, as well as downslope or upslope, from the source, and where the wedge of water and the bottom are isovelocity. The total field in the 3-D wedge is expressed as a sum of ray fields, each of which corresponds to a saddle point of a double integral over plane waves. The broadband case of the benchmarking problem is solved by finding frequency-dependent eigenrays at coarsely spaced frequencies and interpolating the eigenray characteristics. Eigenray diagrams provide an intuitive picture of the horizontal refraction effects associated with three-dimensional underwater acoustic propagation. [Work supported by ONR.]

**2aAOa7. Geoacoustic inversion in range-dependent environments: A plane-wave reflection coefficient approach.** Steven A. Stotts (Appl. Res. Labs., P.O. Box 8029, Univ. of Texas, Austin, TX 78713-8029) and David P. Knobles (Univ. of Texas, Austin, TX 78713)

One of the most challenging aspects of geoacoustic inversion in ocean waveguides involves range-dependent problems, such as in shallow water or continental shelf environments. Inversion calculations may require tens or hundreds of thousands of calls to an acoustic propagation model. When the forward propagation model is range dependent, the computation time can become extremely large. If, however, the range dependence is known, an approximation is introduced that significantly simplifies the problem. Using the measured bathymetry and range-dependent sound-speed profile, the geometrical optics approximation is used to compute the eigenrays between sources and receivers. Assuming a rigid bottom boundary condition, the MEDUSA algorithm is used for this purpose. This range-dependent ray theory calculation separates the propagation in the water column from that in the seabed. The modeled broadband pressure fields are obtained by computing the plane-wave reflection coefficient (PWRC) at the angles and frequencies as needed in the eigenray expansion. Namely, each perturbation of the seabed does not require a new range-dependent calculation of eigenrays, only a recalculation of the PWRC. Examples are shown that demonstrate the accuracy, efficiency, and limitations of the approach. [Research supported by SPAWAR and ONR.]

9:40

**2aAOa8. Theory of acoustic radiation over a 3-D hyperbolic ocean ridge.** Thomas K. Berger (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238)

An analytic solution to a 3-D problem of ocean acoustics involving horizontal refraction is presented. An expression for the acoustic field due to a harmonic point source in an isovelocity ocean overlying a hyperbolic ridge with perfectly reflecting boundaries is given as a finite modal sum of integrals. The integrals are in terms of Mathieu functions, the eigenfunctions of the reduced wave equation in the elliptical coordinate system. These eigenfunctions are approximated away from the low-frequency limit by standard WKB techniques and the resulting integrals are then estimated by first- and second-order stationary-phase asymptotics. This gives an approximation in terms of elliptical functions for which standard numerical routines are available. The approximation is very accurate in the shallow depth and gentle slope limits appropriate to the physical situation of ridges in an isovelocity ocean. The solution includes the locations of the caustics and shadow zones and predicts a complicated intramodal interference pattern resulting from the intersection of up to three rays in a given mode. The intermodal interference from the finite modal sum is also evident in the full solution. The explicit representation of the features arising from horizontal refraction makes this model useful as a 3-D benchmark solution.

9:55–10:10 Break

10:10

**2aAOa9. Application of the enhanced ray theory to benchmark problems.** Nick E. Maltsev (MVM Intl., 10678D Maplewood Rd., Cupertino, CA 95014)

Enhanced ray theory [Maltsev, J. Comp. Acoust. (in press)] was applied to benchmark problems. A new approach for searching of eigenrays, based on the structure of rays surface in phase space, was developed. The sound field is composed by numerous eigenrays, so the time and space structure of the field is preserved and available for interpretation.

**2aAOa10. Broadband PE/SSF modeling in 2-D and 3-D.** F. D. Tappert (Appl. Marine Phys., Rosenstiel School, Univ. of Miami, Miami, FL 33149)

Prediction of pulse propagation for the 2-D and 3-D test cases proposed for benchmarking range-dependent numerical models is accomplished with the  $c_0$ -insensitive version of the broadband parabolic equation with the split-step Fourier (PE/SSF) algorithm numerical model that takes advantage of efficient Fourier syntheses from the frequency domain to the time domain. Special attention is paid to the fast and accurate prediction of the travel times and amplitudes of the resolved multipath arrivals, as well as to the unresolved multipaths. The role of chaos in limiting accurate pointwise predictions in range-dependent (nonseparable) propagation problems is highlighted. [Work supported by ONR Code 3210A.]

10:40

**2aAOa11. Some normal mode and PE propagation predictions for selected benchmark test cases.** A. Tolstoy (ATolstoy Sci., 8610 Battailles Court, Annandale, VA 22003, atolstoy@ieee.org)

Results will be presented for the 3-D and shelf-break test cases. The 3-D problem results here will be for 2-D predictions only. The KRAKEN (adiabatic and coupled versions) and PE (energy conserving and nonconserving) models will be the primary codes investigated. It is found that it is *nontrivial* to run these codes even when one is familiar with them. It is also found that all the models generally agree, but there are some quantitative differences. [Work supported by ONR.]

10:55

**2aAOa12. An implementation of a three-dimensional wave propagation model in a parallel processing computational environment.** Thomas N. Lawrence (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713), Richard D. Pound, and Roy M. Jenevein (Dept. of Computer Sci., Univ. of Texas, Austin, TX 78712-1188)

A parallel processor implementation of a three-dimensional parabolic equation approach to acoustic wave propagation was developed using the University of Texas Cray T3E (Lone Star). Lone Star is currently an 84 processor machine. We will discuss results from modeling various underwater acoustic environments where three-dimensional results are required, and the computation speed-up achieved using a multiprocessor approach. High performance computing, i.e., computation on high speed multiple processor machines, is now entering the lower priced small computer era. Environmental acoustic problems heretofore ignored because of the computational intensity required can now be considered for operational field systems. Although the T3E is a mainframe machine, this model is being developed for a low cost clustered processor machine with a combination of shared and distributed memory, which we expect to have in operation this year. [Work sponsored under Independent Research and Development at the Applied Research Laboratories. Computer support on the T3E is funded by the National Partnership for Advanced Computational Infrastructure through the University of Texas Advanced Computing Center for Engineering and Science.]

11:10

**2aAOa13. Numerical simulations with 3D WAPE considering shallow water range-dependent environments.** Frédéric Sturm (INSA de Lyon, LMC-UMR CNRS 5514 et MAPLY-UMR CNRS 5585, 20 avenue Albert Einstein, 69621 Villeurbanne cedex, France)

A three-dimensional parabolic equation based model, 3-D WAPE, is run considering several shallow water range-dependent environments. This 3-D model is based on a higher-order Padé approximation in depth and a linear approximation in azimuth. Broadband modeling is handled using a Fourier synthesis approach. We focus on the numerical method chosen to solve the inherent initial- and boundary-value problem. When



one is concerned with solving sound propagation problems in any three-dimensional waveguides, one of the main issues is to reduce the azimuthal sampling so as to achieve 3-D computations in a more reasonable time with the same accuracy. The efficiency of using higher-order finite difference schemes in azimuth is discussed. Some numerical results obtained using 3-D WAPE on various test case scenarios are presented, including the ASA 3-D penetrable wedge-shaped waveguide benchmark. For each test case, CPU time comparisons are performed. Another interesting aspect of the marching algorithm used is its ability to be parallelized. Parallel computing appears to be very promising for solving large scale numerical problems. The advantage of using parallel computing is also discussed.

11:25

**2aAOa14. 3-D benchmarking with a three-dimensional azimuthal wide-angle model in the wedge.** Ying-Tsong Lin, Chifang Chen (Dept. of Naval Architecture and Ocean Eng., Natl. Taiwan Univ., 73 Chou-Shan Rd., Taipei, Taiwan, ROC), and Ding Lee (Yale Univ./Naval Undersea Warfare Ctr., CT)

In predicting wave propagation in either direction, the size of the angle of propagation plays an important role; thus, the concept of wide-angle is introduced. Most existing acoustic propagation prediction models do have the capability of treating the wide-angle but the treatment, in practice, is

vertical. This is desirable for solving 2-dimensional (range and depth) problems. In extending the 2-dimensional treatment to 3 dimensions, even the wide-angle capability is maintained in most 3-D models, it is still vertical. This paper uses an azimuthal wide-angle wave equation. The 3-D benchmarking is done by a 3-dimensional azimuthal wide-angle model with real 3-D side-wall boundaries for up-slope calculations. The cross-slope calculations are also done for both subcases. [Work supported by National Science Council of Republic of China.]

11:40

**2aAOa15. A research of sound rays propagated in sloping seabed ocean.** Guanting Jiang and Shengming Guo (Zhongguancun Rd. 17, Beijing, PROC, jgt@oceana.ioa.ac.cn)

The property of sound waves transmitted through sea water always presents an important problem in acoustics. In this paper, sound rays propagated underwater on an irregular seabed were calculated and some interesting results were received. Sound transmitted amplitude is lower than in a plane seabed sea, and in a slope sea bottom the direction of the sound ray will turn deflexion along the bed. All of the results are obtained using the ray-trace method.

2a TUE. AM

TUESDAY MORNING, 5 JUNE 2001

ADAMS ROOM, 8:30 A.M. TO 12:00 NOON

## Session 2aAOB

### Acoustical Oceanography: Acoustical Instrumentation for Water Column Measurements III

Kenneth G. Foote, Chair

*Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, Massachusetts 02543*

Chair's Introduction—8:30

#### Invited Paper

8:35

**2aAOB1. Fisheries applications of Doppler sonar.** Len Zedel (Phys. and Physical Oceanogr. Dept., Memorial Univ. of Newfoundland, St. John's, NF A1B 3X7, Canada, zedel@physics.mun.ca)

Signal processing inherent to broadband Doppler current profiling systems has great potential for applications in fisheries monitoring. The large bandwidth (25%–50% of the carrier frequency) contains information that can be used in target characterization, the system determines current profiles and there is the potential to make direct measurements of fish swimming speed. The capabilities of these systems are explored: field data collected with an RD Instruments 307-kHz Workhorse Acoustic Doppler Current Profiler (ADCP) demonstrate observations of large herring schools, performance is also evaluated using observations made in a tow-tank facility. For large concentrations of fish, speed and direction can be determined but for lower concentrations of fish new data processing techniques are required. For the system configuration as tested, single ping estimates have a standard deviation as low as  $10 \text{ cm s}^{-1}$ . This accuracy is substantially better than the accuracy of  $\approx 40 \text{ cm s}^{-1}$  expected and suggests that for isolated targets the approach is equivalent to fully coherent Doppler processing. There is a difficulty discriminating discrete targets because of the long pulse code sequences transmitted. [This research was funded by the Natural Sciences and Engineering Council of Canada, and by Fisheries and Oceans Canada.]

8:55

**2aAOb2. Inference of size and orientation of swim-bladdered fish from high-frequency acoustic echograms.** Dezhang Chu, Andone Lavery (Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543), and J. Michael Jech (NOAA/NMFS Northeast Fisheries Sci. Ctr., Woods Hole, MA 02543)

It is well known that the volume scattering strength is very sensitive to the size and orientation of scattering objects such as swim-bladdered fish. Such behavioral information is in general not always available. Inadequate knowledge of target orientations limits our ability to correctly estimate the size and abundance of fish acoustically. To extract orientation information of individual fish and/or fish schools, a scattering-model-based image processing method is presented. For fish aggregations within which individual fish can be resolved from the echogram, the geometric shapes and acoustic intensities of the identifiable highlights can potentially be used to infer the sizes and orientations of the individual targets. For fish schools within which individual fish cannot be resolved, the mean fish size and orientation can still potentially be inferred by examining the statistics of the amplitude distribution of the patches resulting from strong backscattering from fish schools. The scattering model is based on the Kirchhoff approximation. The requirements to improve the performance of the presented method and its limitations are discussed.

9:10

**2aAOb3. Calibration and use of a broadband ADCP to measure zooplankton volume scattering strengths.** Mark C. Benfield (Dept. of Oceanogr. and Coastal Sci./Coastal Fisheries Inst., Louisiana State Univ., 218 Wetland Resources, Baton Rouge, LA 70803, mbenfie@lsu.edu), Duncan E. McGehee (Ocean Sci. Res. Group, BAE Systems, San Diego, CA 92123-4333), Sean F. Keenan (Louisiana State Univ., Baton Rouge, LA 70803), and Kent L. Deines (DrDoppler.com, Poway, CA 92064)

A 1200 kHz broadband ADCP from RD Instruments was calibrated and used to collect quantitative volume backscattering data from zooplankton in the Gulf of Mexico. Voltages on the four receivers, reported in terms of "counts," were linear with respect to received sound pressure level in dB *re* 1  $\mu$ Pa, with sensitivities of 2.376 to 2.434 counts/dB, depending on the receiver. Source levels ranged from 212.9 to 214.6 dB *re* 1  $\mu$ Pa at 1 m. The ADCP was mounted on the C/V Admiral Semmes and used to survey volume backscattering around an oil platform in the Gulf of Mexico. Volume backscattering strengths measured between -40 and -90 dB, which were generally consistent with independent observations of the numbers and sizes of scatterers present in the water column. These results suggest that, with careful calibration, ADCPs may be used for quantitative zooplankton research.

9:25

**2aAOb4. New system for remotely monitoring the three-dimensional movement of acoustically tagged fish.** Tracey W. Steig, Samuel V. Johnston, and Bruce H. Ransom (Hydroacoustic Technol., Inc., 715 NE Northlake Way, Seattle, WA 98105, support@htisonar.com)

A passive acoustic tag system was developed to monitor the three-dimensional movements of migrating fish with submeter resolution. The acoustic tag receiver monitors an array consisting of up to 16 omnidirectional hydrophones, with received signals synchronized to determine the arrival times for each pulse transmitted by the acoustic tag. Arrival times are then used to calculate the three-dimensional position of a tagged fish as it moves through the array. Algorithms were developed to precisely calculate the three-dimensional positions of the hydrophones, and of each acoustic tag. Over the last 4 years, this system was used at several dams in the United States. Most studies to date monitored downstream migrating juvenile salmonids as they approached and passed turbine intakes, spillways, and juvenile bypass systems at hydroelectric dams. Fish movement patterns were tracked in three dimensions over time, typically with sub-meter resolution. Tagged fish were 160–240 mm long. Acoustic tags were approximately 7 mm in diameter by 23 mm long, weighted 2 g, and

transmitted at 307 kHz. Tag codes (up to 500), pulse width (typically 1–5 ms), and ping rate (typically 0.3–3 pings/s) were field programmable. Current tags incorporate signal encoding for an improved signal-to-noise ratio, and weigh as little as 1 g.

9:40

**2aAOb5. Acoustically characterizing detritus with a 420-kHz echosounder.** Diane E. Di Massa<sup>a)</sup> (Univ. of Massachusetts–Dartmouth, II-116A, 285 Old Westport Rd., North Dartmouth, MA 02747, ddimassa@umassd.edu)

This paper presents the new application of hydroacoustically measuring and characterizing detritus (here defined as suspended decomposing plant material) for shallow-water environments such as bays, estuaries, or rivers. Detrital material was measured hydroacoustically using a 420-kHz digital echosounder, first in a controlled laboratory test and then in an open estuary. Controlled tests on both small matted masses and individual grass strands produced signals clearly visible (6–15 dB) above the background noise. Field tests were performed using ship-mounted downward looking sonar and echo integration software to determine the feasibility of using hydroacoustics to measure, characterize, and potentially quantify the detrital biomass. A vertical series of plankton nets, slightly downstream of the echosounder, and flow velocity measurements by a 1200-kHz acoustic doppler current profiler (ADCP) slightly upstream provided ground-truth information of local concentrations. Hydroacoustics proved successful in identifying variable material concentrations both spatially and temporally, with typical backscatter intensities of -70 dB. Echo amplitude information from the ADCP showed a first-order correlation with backscatter detected by the echosounder and with net sample volumes. <sup>a)</sup>Also affiliated with Woods Hole Group, 81 Technology Park Drive, East Falmouth, MA 02536.

9:55

**2aAOb6. The new Simrad EK60 scientific echo sounder system.** Lars Nonboe Andersen (Simrad AS, P.O. Box 111, N-3191 Horten, Norway)

A widely used tool for fish abundance estimation and other marine research is the scientific echo sounder. Acoustic information obtained from scientific echo sounders such as target strength and target position can be valuable for describing individual targets whereas measurements of volume backscattering strength are suitable for describing a large number of targets distributed over a volume of water. The new Simrad EK60 scientific echo sounder system provides high-quality target strength, target position, and volume backscattering strength measurements over a large dynamic range for a wide range of frequencies (18–200 kHz). An overview of the EK60 will be presented and operation of the echo sounder will be demonstrated. The presentation will include demonstration of the calibration procedure of the EK60.

10:10–10:20 Break

10:20

**2aAOb7. A quantitative echo sounder with multiple beams, model KFC-3000.** Kenji Kuratsu, Takahiro Satho (Kaijo Corp., 3-1-5, Sakae-cho, Hamura-shi, Tokyo 205-8607, Japan), Yoichi Miyanoana, Kouichi Sawada, and Yoshimi Takao (Natl. Res. Inst. of Fisheries Eng., Ebaidai, Hasaki, Kashima-gun, Ibaraki 314-0421, Japan)

We have developed a quantitative echo sounder with various features such as effective combination of dual beam and split beam methods. This system is designed in accordance with the established designing method of a quantitative echo sounder. The method specifies source power, operating frequency, and so on for a given measuring object and range under the principle of minimum estimation error. Ideal omnidirectional and symmetrical simultaneous split beam methods detect the three-dimensional position of an individual target. An omnibus display on the multiwindows

provides full echo information. Raw data for post-processing is used for accurate biomass evaluation. The easy operation of this system offers a wider range of users more adequate and reliable data for fishery resource surveys.

10:35

#### 2aAOB8. Hydroacoustic performance of fishery survey instruments.

Hans P. Knudsen, Egil Ona, and Ingvald Svellingen (Inst. of Marine Res., P.O. Box 1870, Nordnes, N-5817 Bergen, Norway)

Accurate estimation of fish stock abundance by acoustic surveying methods is now essential for long-term assessment of many of the world's largest fish stocks and the status of their ecosystems. Significant improvements of the acoustic oceanic surveying methodology have been achieved over the last decade, in particular in calibration stability, receiver design, post-processing systems and transducer installations for oceanic, bad-weather operation. Experiences with introducing these new elements on four ocean-going research vessels, with examples, are presented and discussed. Since many of the potential survey errors related to the technical instrumentation seem to be overcome or minimized, the present focus for improvements must now be on biotic factors such as target strength variability, vessel avoidance and vertical distribution of the fish relative to the acoustic sampling volumes. The effect of significantly reducing vessel-emitted noise and improved observation methods in the echo sounder dead zones are discussed.

10:50

#### 2aAOB9. Detection range of acoustic instruments for fisheries.

Masahiko Furusawa, Jusam Park, Myounghee Kang, and Chumming Fan (Tokyo Univ. of Fisheries, Tokyo, Japan)

Detection ranges of acoustic instruments mainly used for fisheries and their research are derived as the range bordered by a certain signal-to-noise ratio (SNR) threshold. The SNR is depicted by several factors on transmitting and receiving, sound propagation, scattering by objects, and

mainly self-ship noise. The detection ranges are shown for several fisheries instruments: fisheries echo sounder, quantitative echo sounder, and bio-telemetry system. The results can be used for designing the instruments, examining the capability of user's own instruments, and interpreting obtained data or echograms. Some examples of the results follow. Increasing transmitting power is not as effective for high frequencies as for low frequencies to increase the detection range. Comparison of volume backscattering strengths obtained by the quantitative echo sounder at several frequencies for the purpose of rough species' identification should be done within the same detection range. By applying the concept of the detection range for the bio-telemetry receiver beams, the number of the beams and the beamwidths can be determined.

11:05

#### 2aAOB10. Experimental tests of a new method to monitor sea medium by analyzing ambient noise and reradiating it from a distant point.

V. Furduev Alexander (Head of Lab. Andreyev Acoust. Inst., 4 Shvernik Str., 117036 Moscow, Russia) and D. Svet Victor (Head of Lab. Andreyev Acoust. Inst., 117036 Moscow, Russia)

In-sea experiments show a possibility to estimate parameters of the water column by analyzing space-time characteristics of the natural ambient noise. If the noise is reradiated by sound transponder located at a distance from the receiver, a regenerative monitoring scheme can be implemented. In such scheme a feedback is present that increases the contrast of the spectrum maximums to higher accuracy. By increasing the amplification gain of the whole circuit (including the feedback loop of the underwater channel), one comes to a self-sustained oscillator, its frequency deviations are indicating variations of the medium parameters. Experimental examples are presented to confirm feasibility of the proposed monitoring technique for measuring temperature variations, current velocity, periods of internal waves, and other features of the water column. The method is especially applicable to small sea areas like straits, harbors, and lakes to monitor their environmental equilibrium or manifestations of human activity. [Work supported by RFBR, Project No. 00-05-64226.]

11:20–12:00

#### Panel Discussion

TUESDAY MORNING, 5 JUNE 2001

SALONS III/IV, 8:00 TO 9:15 A.M.

#### Session 2aBBa

### Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound and Vibration in Musculoskeletal Structures

R. Glynn Holt, Chair

Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

#### Contributed Papers

8:00

#### 2aBBa1. Evaluation of vibratory coherence as an alternative to radiography in assessing bone healing after osteo-distraction.

Tarek H. El-Bialy, Thomas J. Royston, and Akira Sakata (Univ. of Illinois at Chicago, Chicago, IL 60607, troyston@uic.edu)

Distraction osteogenesis is used in orthopedics to lengthen bones, by cutting or breaking the bone and gradually separating the two pieces as new bone fills the intervening space. There is a need for early assessment of the degree of bone healing that allows for normal functioning without unwanted side effects. This study compared different techniques used to evaluate the degree of bone healing during mandibular osteodistraction in

21 rabbits. For each rabbit, the mandible was cut in a surgical procedure and then 72 h later distraction began at a rate of 3 mm per day. Bone formation at the distraction site was assessed by *in vivo* photodensitometry on head radiographs, an *in vivo* (nondestructive) vibratory coherence test across the distraction site, a post-mortem, *ex vivo* (destructive) three-point bending mechanical test, and by post-mortem, *ex vivo* (destructive) histological examination. Statistical analyses included analysis of variance (ANOVA) and correlation coefficient tests. The findings revealed that the results of bone photodensity and the vibration technique are positively correlated to the results of the mechanical three-point test and histological examination. The use of the vibration technique may provide a substitute for or augment the routine use of radiography for *in vivo* evaluation and monitoring of bone healing.

**2aBBa2. Physical mechanisms implied in ultrasound propagation through trabecular bones.** Frederic Padilla and Pascal Laugier (Laboratoire d'Imagerie Parametrique, P6-CNRS UMR 7623, 15 rue de l'Ecole de Medecine, 75006 Paris, France, padilla@lip.bhdc.jussieu.fr)

Measurements of quantitative ultrasound parameters represent an established mean of skeletal status assessment in osteoporosis. However, to date, the exact physical mechanisms underlying ultrasound attenuation in cancellous bone have not been clearly documented. This study reviews the different mechanisms underlying propagation of ultrasound in trabecular bone, based on results obtained by our group and others. Bone is a porous, anisotropic, and scattering medium. Anisotropy of phase velocity has been explained with a multilayer model. Several attempts have been made to model the propagation with the Biot's theory for poroelastic media. Encouraging results have been obtained for velocity. However, the viscous absorption mechanism implied in this theory disagrees with the experimental attenuation data and with the observed influence of fluid viscosity. Negative dispersion has been recorded, in agreement with some multiple scattering ideas. Several models of scattering (analytical and numerical) have given good predictions of backscattering coefficient. Despite the good results observed with these models, none of these could explain the experimental data, and especially the attenuation data. In conclusion, a more complex model will be discussed, to describe both absorption and scattering effects, as in geophysics for rock studies.

**2aBBa3. Measurements of attenuation and backscattering in trabecular bones over a large frequency range.** Frederic Padilla, Sana Chaffai, Genevieve Berger, Pascal Laugier (Laboratoire d'Imagerie Parametrique, P6-CNRS-UMR 7623, 15 rue de l'Ecole de Medecine, 75006 Paris, France, padilla@lip.bhdc.jussieu.fr), and Francoise Peyrin (CREATIS, UMR INSA 502, 69621 Villeurbanne Cedex, France)

In order to understand the propagation of ultrasound through trabecular bones, measurements of attenuation and backscattering coefficients were performed. Attenuation was measured on 14 human calcanei, over a large frequency bandwidth (0.2–1.7 MHz). The experimental attenuation coefficient values were modeled with a nonlinear power fit  $\alpha(f) = a_0 + a_1 f^n$ . The attenuation coefficient was found to increase as  $f^{1.09 \pm 0.3}$ . A highly significant relationship was noted between  $a_1$  and BMD. No correlation was found between  $n$  and BMD. The backscatter coefficient was measured in 19 bones specimen in the frequency range 0.4–1.2 MHz. The experimental frequency dependence was found to be  $f^{3.4 \pm 0.3}$ . A twofold theoretical approach was then adopted. The analytical model of Faran for spherical and cylindrical elastic cylinders showed a qualitative agreement with experimental data. A better agreement was found with a weak scattering medium model, where the backscatter coefficient is related to the autocorrelation function of the propagating medium. These results first open interesting prospects for the investigation of the influence of bone microarchitecture on ultrasonic scattering; second it seems to indicate that scattering is not the main mechanism of ultrasonic attenuation, because of the two different frequency dependencies, as confirmed by a numerical fit (absorption + scattering).

**2aBBa4. Cortical bone characterization by guided waves.** Maryline Talmant, Emmanuel Bossy, Estelle Camus, Frederic Padilla (LIP, UMR CNRS 7623, 15 rue de l'Ecole de Medecine, 75006 Paris, France), and Pascal Laugier (LIP, UMR CNRS 7623, 75006 Paris, France, talmant@lip.bhdc.jussieu.fr)

The axial transmission method was first developed in 1958 to study the status of cortical bone during fracture healing. It has been subsequently used to investigate bone strength in osteoporosis. Two transducers are placed on the same side of the bone and the velocity of the first arriving signal on the receiver is measured. The present paper contributes to the understanding of the type of propagation involved in this technique when applied on bone. When using a propagation model for semi-infinite media the first signal is identified as the longitudinal lateral wave for a defined arrangement of transducers. However, the ratio of bone thickness over wavelength is of the order of unity and then normal modes of vibration give a non-negligible contribution to the reflected pressure. Whereas a lateral wave investigates the bone superficially, shell modes penetrate inside the material much more. Signal simulations, *in vitro* experiments and *in vivo* measurements using our prototype will be presented. Particular attention will be given to the influence of cortical thickness. The velocity of the first signal decreases with respect to the thickness, from the velocity of the lateral wave to the velocity of the extensional mode of the lowest order.

**2aBBa5. Ultrasonic pulse transmission technique to evaluate biomechanical properties of ACL graft substitutes.** Krishna V. Kambhampati (Dept. of Bioengineering, Aurobindo Pharma Ltd., Plot #2, Maithri Vihar, Ameerpet, Hyderabad, 500 038, India, rad@aurobindo.com)

The study aims at the identification of an ideal graft material for arthroscopy guided anterior cruciate ligament (ACL) reconstruction (ACLR). Ultrasonic pulse transmission technique was used to analyze the elastic properties (EP) of fresh human ACL, semitendinosus tendon (ST), middle one-third of patellar tendon (CPT) and cigarette rolled, distally based iliotibial band (CITB) grafts in an untwisted and incremented number of twists (two to eight times) at 1 MHz fundamental frequency. The results indicated that the strength of the graft depended upon its collagen density and the EP depended upon the pretensioning of the graft for ACLR. The EP were shown to have significant difference on selective pretensioning of the graft. The EP of CITB grafts changed after twisting. Rigidity of twisted CITB graft surpassed that of the ACL, ST, and CPT grafts. The CITB graft twisted five to seven times along its long axis is isoelastic with normal ACL and CITB graft twisted eight times is stronger than even CPT graft. Twisting of CITB graft provides initial strength and stiffness, necessary pretensioning and the balance between elasticity and rigidity is achieved to duplicate the EP of ACL. The success of ACLR depends upon using an isoelastic graft taken from a relatively, less important CITB.



## Session 2aBBb

## Biomedical Ultrasound/Bioresponse to Vibration: Topics in Medical Acoustics

Paul L. Carson, Chair

*Department of Radiology, University of Michigan Medical Center, Kresge III, R3315, 200 Zina Pitcher Place,  
Ann Arbor, Michigan 48109-0553*

## Contributed Papers

9:45

**2aBBb1. Active ultrasound imaging of breast tumors: Forward and inverse scattering methods.** Yanqing Zeng, Zhongqing Zhang, and Qinghuo Liu (Dept. of Elec. and Computer Eng., Duke Univ., Box 90291, Durham, NC 27708-0291, qhliu@ee.duke.edu)

Ultrasonic imaging for the detection of breast tumors is an important technique to complement existing x-ray mammography. The potential advantages of the ultrasound imaging technique stems mainly from the relatively high contrast of acoustic properties between tumors and normal breast tissue. However, this high contrast also increases the difficulty of forming an accurate image because of the increased multiple scattering. To address this issue, we have developed fast forward methods based on a combination of extended Born approximation, conjugate- and biconjugate-gradient methods, and fast Fourier transform; we propose two nonlinear ultrasound imaging algorithms to improve the resolution for the high-contrast media encountered in ultrasound breast tumor detection. The measured data are simulated with a forward method. The inverse scattering algorithms account for the complicated multiple scattering (diffraction) effects. Numerical results show that our algorithms can accurately model and invert the high-contrast media in breast tissue. The outcome of the inversion is a high-resolution digital image containing the physical properties of the tissue and potential tumors. Compared with a conventional x-ray mammography, the ultrasound imaging system with an accurate imaging algorithm has the potential to provide better specificity between benign and malignant tissues.

10:00

**2aBBb2. Modeling sound transmission through the pulmonary system and chest with application to diagnosis of a collapsed lung.** Xiangling Zhang, Thomas J. Royston (Univ. of Illinois at Chicago, Chicago, IL 60607, troyston@uic.edu), Hussein A. Mansy, and Richard H. Sandler (Rush Medical College, Chicago, IL 60612)

A study was undertaken to demonstrate the feasibility of using audible-frequency vibro-acoustic waves for diagnosis of pneumothorax, a collapsed lung. The hypothesis was that the acoustic response of the chest to external excitation would change with this condition. External acoustic energy was introduced into the trachea via an endotracheal tube. For the control (nonpneumothorax) state, it is hypothesized that sound waves primarily travel through the airways, couple to the lung parenchyma, and then are transmitted directly to the chest wall. In contradistinction, when a pneumothorax is present the intervening air presents an added barrier to efficient acoustic energy transfer. A theoretical model of sound transmission through the pulmonary system and chest region to the chest wall surface is developed to more clearly understand the mechanism of intensity loss when a pneumothorax is present, relative to a baseline case. The model predicts a decrease in acoustic transmission strength of as much as two orders of magnitude when a pneumothorax is present. This is approximately in agreement with experimental measurements on mongrel canines. Development of the model and its comparison with experimental canine studies will be reviewed. [Research supported by NIH NCRR Grant No. 14250 and NIH NHLBI Grant No. 61108.]

10:15

**2aBBb3. Detection of ultrasound generated contrast bubbles in a refluxing canine model.** Emma Y. Hwang, J. Brian Fowlkes, Paul L. Carson (Dept. of Radiol., Univ. of Michigan, Kresge III, R3315, Ann Arbor, MI 48109-0553), and David A. Bloom (Univ. of Michigan, Ann Arbor, MI 48109-0330)

A surgical reflux procedure, performed on two canines, was used to test a less invasive method for diagnosing urinary reflux. After approximately 6 weeks of recovery, examinations tested for reflux over the next 7 weeks. The canines were positioned prone (slightly head down) on a tilted table with a hole exposing the depilated abdomen, where a 1.18 MHz high intensity focused ultrasound transducer could be coupled. The reflux condition was evaluated first using standard fluoroscopy and x-ray contrast instilled in the bladder by a gravity feed through a transurethral catheter. For high grade reflux, dilute contrast agent (Optison) was tested using harmonic color Doppler at 3 MHz on a Toshiba Powervision 8000 scanner imaging the kidney. Finally, aqueous solutions with elevated CO<sub>2</sub> were placed in the bladder and microbubbles generated using a 10 s application of 1.18 MHz ultrasound (25 cycle bursts with 0.5% duty factor and 8.4 MPa). Bubbles were detected in the kidney on two occasions with strong color Doppler signal, near that from dilute Optison. Methods can naturally produce elevated CO<sub>2</sub>, and contrast bubbles should be producible transcutaneously for reflux diagnosis without ionizing radiation and catheterization. [Work supported in part by a USPHS grant 2 RO1DK42290.]

10:30

**2aBBb4. High frequency backscatter and attenuation measurements of porcine red blood cell suspensions between 30 and 90 MHz.** Subha Maruvada (Focused Ultrasound Surgery Group, Dept. of Radiol., Brigham and Women's Hospital, 221 Longwood Ave., LMRC 13, Boston, MA 02115), K. Kirk Shung, and Shyh-Hau Wang (The Penn State Univ., University Park, PA 16802)

There are now diagnostic ultrasonic imaging devices that operate at very high frequencies (VHF) of 20 MHz and beyond for clinical applications in ophthalmology, dermatology, and vascular surgery. To be able to better interpret these images and to further the development of these devices, knowledge of ultrasonic attenuation and scattering of biological tissues, such as blood, in this high-frequency range is crucial. VHF experiments were performed on porcine red blood cell (RBC) suspensions, which have been used in ultrasound experiments by many investigators but in a lower frequency range. Attenuation and backscatter from various hematocrit levels (6%, 10%, 15%, 20%, 25%, 30%) from 30 to 90 MHz were measured using focused transducers. The dependence of the attenuation coefficient from all suspensions followed linearly while the backscatter coefficient for low hematocrit suspensions was found to be a maximum between 10% and 15%. The higher hematocrits showed a decrease in their frequency dependence, possibly no longer indicating Rayleigh scattering since the wavelength is not much larger than the size of a porcine RBC. These results as well as the frequency limits of these type of scattering experiments will be discussed.

**2aBBb5. Scattering by a red blood cell: the importance of particle shape and orientation.** Constantin-C. Coussios and John E. Ffowcs Williams (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, UK, ccc20@eng.cam.ac.uk)

Red blood cells, which normally present themselves as biconcave discs, have traditionally been modeled as spherical scatterers of equivalent volume. The simplicity and symmetry of the spherical model does not however account for the effect of the angle of incidence of the incident field onto the red blood cell. Intuition tends to suggest that the field scattered by an asymmetric particle would differ if the particle were impinged upon broad-side or thin-side, an effect that is likely to become important if a collection of scatterers are aligned in any particular plane. Our objective is to compare the scattering by a red blood cell *of given volume* modeled as a sphere, to the result obtained by approximating its shape as a disc of varying aspect ratio. Results strongly suggest that the spherical model provides a good description for frequencies up to 20 MHz, beyond which particle shape becomes important. Although this effect could go undetected if the scattered field is only measured at angles of 180° (backscattering) and 90° relative to the direction of the incident field, the importance of particle shape and orientation might nevertheless form the base of a novel detection technique.

**2aBBb6. Ultrasonic characterization of proteins in complex fluids.** Donato Valdez, Marcel Gindre, Jean-Yves Leherou, Marcel Waks (Laboratoire d'Imagerie Parametrique, 15 rue de l'école de médecine, F-75006 Paris, France), and Wladimir Urbach (Laboratoire de Physique Statistique, Paris, France)

Our aim is the study of the physicochemical protein properties (volume, compressibility) as a function of hydration by an ultrasonic technique. For this purpose, we have used a biomimetic medium: reverse micelles, a system where we can control precisely water concentration. The compressibility of a medium is obtained by measuring the density and the ultrasonic celerity. The difficulty is increased in a complex fluid: na-

nometric size inclusions (micelles) dispersed in an organic solvent. To reach this goal, we have custom-built in our laboratory an apparatus allowing the determination of ultrasonic celerity, in very small volumes, with a relative precision of about 10 ppm. Using the effective medium theory, we have determined the compressibility of these inclusions with a precision better than 1%. With a spherical model for micellar inclusions and with the hydrodynamic radius of micelles obtained by x rays, we have estimated the compressibility of the water within micelles. We have also evidenced a difference in compressibility between the central core water and water bound to the polar surfactant headgroups. Finally by applying the mean field theory, we have obtained an estimate of the compressibility of proteins inserted within the micelles, at various hydration levels.

**2aBBb7. A hybrid model to simultaneously determine ultrasound wave velocity and thickness of multilayered media.** Ana V. G. Sousa, Wagner C. A. Pereira, and Joo C. Machado (Biomed. Eng. Prog., COPPE, Federal Univ. of Rio de Janeiro, P.O. Box 68510, Rio de Janeiro, Brazil 21945-970)

This work presents a method to simultaneously obtain the wave velocity ( $wv$ ) and layer thickness ( $h$ ) in a stratified media, in order to generate a parametrical image. The influence of refraction is also measured and included in the propagation model. In biomedical ultrasound (US), this certainly would help diagnostic procedures, as it identifies different tissues that conventional US cannot discriminate, like benign cysts from malignant tissue. It can also be applied to determine the degree of corrosion of metal ducts and delamination of aircraft painting. The model is based on geometrical acoustics and uses two transducers to obtain experimental data (256 echoes per layer): one active (3.4 MHz) and a hydrophone, which is moved laterally through 15 positions and rotated. A two-layered phantom (alcohol 6.2 mm and acrylic 6.0 mm) is immersed in a water tank andinsonified. The echoes are captured and processed using a cross-correlation-based method. The water layer parameters ( $wv$  and  $h$ ) have an accuracy of 1% rms and 2% for precision; alcohol layer, 4% and 9%; and acrylic layer, 12% and 15%, respectively. The experimental results demonstrate that the model has potential to be investigated and will be applied to *in vitro* data.

TUESDAY MORNING, 5 JUNE 2001

PDR 18, 8:00 TO 11:10 A.M.

## Session 2aEA

### Engineering Acoustics and Underwater Acoustics: Transducers for Underwater Vehicle Sonars

Thomas R. Howarth, Chair

NAVSEA Newport, 1176 Howell Street, Building 1171, Code 20, Newport, Rhode Island 02871

Chair's Introduction—8:00

#### Invited Papers

**2aEA1. New sonar concepts and the sensor-related developments they require.** Brian Houston and Joseph Bucaro (Naval Res. Lab., Code 7130, 4555 Overlook Ave., SW, Washington, DC 20375-5350)

Recently, exciting new approaches to the underwater vehicle acoustic sensing problem have been proposed which, if realized, could revolutionize how we approach underwater sonar. Two such schemes are the so-called "virtual sonar" which is formulated around Helmholtz integral processing and "wireless" systems which transfer sensor information through radiated rf signals. The full implementation of such new conceptual approaches, while presenting exciting possibilities, will nevertheless require a number of sensor-related technology advances. In this presentation, we will discuss those sensors and sensor-related developments which are desired or required in order to make practical such new sensor system concepts, and we will indicate the status of such developments. Finally, we will present several underwater applications from the perspective of exploiting these new sonar concepts. [Work supported by ONR.]

8:30

**2aEA2. Broadband transducer requirements and development for a synthetic aperture sonar.** Jose E. Fernandez, James T. Christoff, Kerry W. Commander, and Daniel A. Cook (Coastal Systems Station, Code R21, 6703 W. Hwy. 98, Panama City, FL 32407-7001, fernandezje@nsc.navy.mil)

Broadband acoustic signal transmission and reception are key to enhancing the performance of underwater acoustical sensor systems to support functions like underwater acoustic communications and high resolution underwater imaging. In particular, synthetic aperture sonar (SAS) is an attractive technology for high-resolution imaging because of its ability to produce better cross range resolution than that of a real aperture array sonar of the same length. In addition, SAS systems can produce a cross range resolution that is independent of frequency and range. When the advantages of a SAS system are coupled with those gained from going to a broadband signal space, the performance enhancements can be significant. Specifically improvements in clutter rejection, object classification, and area coverage rate will be achieved. A description of the transducer requirements and expected payoffs for a SAS system designed to operate and image proud and buried objects in shallow waters will be presented. [Work supported by ONR.]

8:55

**2aEA3. Cymbal panels: Low-frequency acoustic projectors for underwater vehicle SONARS.** James F. Tressler (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375) and Thomas R. Howarth (Naval Undersea Warfare Ctr., Newport, RI 02841)

The Naval Research Laboratory is developing low frequency SONAR systems for an unmanned underwater vehicle (UUV) for shallow-water applications. Included among the onboard SONAR systems are three thin, lightweight, low-volume occupation acoustic sources known as cymbal panels. Each panel features miniature Class V flexensional driver elements, called cymbals, sandwiched between two stiff, lightweight radiating plates where each of the three panels is designed with a tailored cymbal element parameter in order to cover a different frequency band. NRL is placing each of these panels next to each other on the UUV in order to cover the desired frequency range from 1 kHz to over 6 kHz. A duplicate set of three panels is also placed on the opposite side of the UUV to provide some directivity means. This presentation will provide an overview of the program objectives, the acoustic source objectives, the element and panel designs, and recent underwater electroacoustic calibration data. [Work supported by Jan Lindberg of the Office of Naval Research Code 321SS and Bruce Johnson of the Office of Naval Research Code 321TS.]

9:20

**2aEA4. Wideband 1–3 piezoelectric composite parametric mode projectors for UUV/AUV applications.** Kim C. Benjamin (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841)

Unlike conventional parametric mode projectors, the 1–3 piezoelectric composite-based parametric mode projector affords a broad frequency range of operation. Due to its intrinsically low mechanical quality factor,  $Q_m$ , the 1–3 piezoelectric composite material allows a wide range of primary frequencies to be used to excite the parametric mode effect. Furthermore, unlike conventional parametric mode projectors, the 1–3 piezoelectric composite substrates do not require inherently narrow band tuning circuits to enhance their half-wave resonance frequency. This presentation will report on the design and measured results for two 1–3 piezocomposite-based parametric mode projectors. The first case consisted of a 23-mm-diam by 6-mm-thick active layer driven with primary frequencies centered about 200 kHz. Beamwidths of 3.5 to 4.0 degrees were realized at difference frequencies ranging from 5–50 kHz. In the second case, a 15-mm by 15-mm by 25-mm-thick active layer was driven with primary frequencies centered about 60 kHz. In this case beam widths of 8.5 to 9.0 degrees were realized at difference frequencies ranging from 1 to 10 kHz.

9:45–10:00 Break

10:00

**2aEA5. Geometrically phased “doily” array.** W. Jack Hughes (Appl. Res. Lab., Box 30, State College, PA 16804, wjh2@psu.edu)

The theory and development of geometrically phased arrays will be presented. The array has evolved from a multibeam PVDF receiver, where its shape looks like a “doily,” to a 1–3 composite pulse-echo imaging array that can be a 1-D or 2-D array capable of 3-D imaging. The main advantage of the doily array is that it can form a set of steered beams without the need for electronically applying phase shifts, drastically reducing the cost and complexity of a system. The steering angle is a function of frequency, allowing the design of multiple steered beams or an imaging array if an FM sweep or large bandwidth pulse is used. In a manner similar to geometric shaping of a transducer for low side lobes, to form steered beams the transducer uses an amplitude function shaped as a cosine and sine aperture, shifting one signal 90 deg. Several arrays have been fabricated using electroplated and etched electrodes on piezoelectric materials such as PVDF and 1–3 composite materials. Beam formation and imaging results will be shown. [Work supported by ONR and DARPA.]

2a TUE. AM



10:25

**2aEA6. Design of a steerable parametric-mode array.** Steve Forsythe and Kim Benjamin (Naval Undersea Warfare Ctr., 1176 Howell Ave., Newport, RI 02871)

A design is presented for a steerable parametric array that operates in the range of 40–60 kHz. The array design is based on three technologies: (1) Parametric array technology: the nonlinear interaction of two narrow high-frequency beams used to create a virtual endfire array with a very narrow beam at a much lower frequency; (2) Frequency-steered array design based on the ideas developed at Penn State University [Hughes and Thompson, *J. Acoust. Soc. Am.* **59**, 1040 (1976)]; (3) Use of 1-3 piezocomposite as the active element of the array. The array is steered in two directions with minimum complexity of the required drive electronics. Steering along one axis is done by a simple change of primary frequencies, while steering along the other axis uses time-delay between elements. This allows the resulting parametric beam to be steered independently in two directions. The array is designed by discretizing the real and imaginary parts of the inverse DFT of the desired far-field patterns of the primaries. Scaling the 2-D DFT of the discretized array pattern allows an easy verification of the final array performance. Implementation issues, such as efficient element interconnection, drive electronics, and fabrication, will also be discussed.

10:40

**2aEA7. Forward-looking acoustic mine sensing.** R. Daniel Costley (Miltec, Inc., Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677), James M. Sabatier, and Ning Xiang (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

A mine sensing system based on acoustic-to-seismic coupling sonifies the ground with high amplitude (120 dB), broadband (80–300 Hz) sound and measures the resulting ground vibration with a scanning laser Doppler vibrometer (LDV). Images produced by these scans show a distinct contrast in several frequency bands between ground vibrations over a buried mine and not over the mine. In order to reduce the risk to the vehicle and

its operators and to allow a safe stopping distance for the vehicle, it is desirable that the system looks a significant distance in front of the vehicle. Experiments were conducted to investigate the feasibility of a forward-looking system. In a forward-looking system, both the sound source and the LDV are moved farther from the scanned area. This configuration both reduces the sound pressure level at the scanned area and decreases the angle at which the LDV beam strikes the ground. These effects reduce the contrast between the over-mine and off-mine signals. In addition, the image is distorted at the shallower LDV-ground angles. Results from these experiments will be presented and discussed. [Work supported by U.S. Army Communications-Electronics Command, Night Vision and Electronic Sensors Directorate.]

10:55

**2aEA8. Acoustic intensity scattered from an elliptic cylinder.** Kang Kim and Gerald C. Lauchle (Grad. Prog. in Acoust. and Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

A computational study is described where a 2-D elliptic cylinder is insonified by a plane acoustic wave. The elliptic cross section of the cylinder has a fineness ratio of 5:1, the incidence angle of the plane wave is 30 and 60 degrees relative to the major axis of the ellipse, and  $ka=20$ , where  $a$  is the major axis of the elliptic cross section and  $k$  is the acoustic wave number. The calculations are performed using the finite element method of solution for partial differential equations. The MATLAB formulation solves the Helmholtz equation with reflection-free conditions imposed on the computational outer boundary, and rigid conditions imposed on the surface of the scatterer. Of particular practical interest in this study is the spatial distribution of the total active acoustic intensity, i.e., the sum of the incident and scattered intensity components. Active intensity amplitude, and the phase between pressure and particle velocity, are computed and compared to pressure amplitude only. The results show that there is significant phase distortion in the forward scattered direction that could be useful in localizing targets in active bi-static operations if  $p-u$ -type acoustic intensity probes were employed. [Work supported by ONR, Code 321 SS.]

TUESDAY MORNING, 5 JUNE 2001

SALONS V/VI/VII, 10:00 A.M. TO 12:30 P.M.

## Session 2aEDa

### Education in Acoustics: Hands-on Demos with High School Students

Uwe J. Hansen, Chair

*Department of Physics, Indiana University, Terre Haute, Indiana 47809*

This session is intended to communicate the excitement of “doing” acoustics to high school students in the Chicago area. Individual demonstrations with associated instrumentation will be presented in a brief introduction. In the subsequent hands-on session, presenters will guide individual students to discover the fun of acoustics by letting them “twiddle the knobs.” Topics available for student participation will range in sophistication from simple wave observation to normal mode mapping on musical instruments. While attendance as well as participation by conference attendees is welcome, hands-on participation is primarily designed for the students attending this session by special arrangement.

## Session 2aEDb

## Education in Acoustics: Undergraduate Research Poster Session

E. Carr Everbach, Chair

*Department of Engineering, Swarthmore College, 500 College Avenue, Swarthmore, Pennsylvania 19081-1397*

## Contributed Papers

All posters will be on display from 11:00 a.m. to 2:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 11:00 a.m. to 12:00 noon and contributors of even-numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m.

**2aEDb1. Pipe impedance and sound spectrum measurements of the khaen and the sheng.** Daniel R. Hoover and James P. Cottingham (Phys. Dept., Coe College, Cedar Rapids, IA 52402, drhoover@coe.edu)

Although the Asian free reed mouth organs all employ metal free reeds mounted in resonating pipes, there are significant differences among different families of these instruments. The sheng employs the free reed at the end of a closed pipe resonator with both conical and cylindrical sections. The khaen pipe has an unusual configuration for a wind instrument, with the free reed placed at approximately one-fourth the length of an open cylindrical pipe. Impedance curves have been measured for several pipes from a sheng and for artificial khaen tubes constructed from PVC pipe. These are compared with the sound spectra produced when the pipes are played. The sounding frequencies for both types of pipes are typically found to be above the natural frequency of the reed and close to, but slightly above, the frequency of the first impedance peak of the pipe. For khaen pipes constructed with the fundamental frequency of the pipe substantially below that of the reed, the sounding frequency is close to the second harmonic of the pipe. These results are consistent with those of some earlier studies as well as with theoretical considerations.

**2aEDb2. Using your ears to learn about the acoustics of metal tubes.** Konrad Kaczmarek (Yale Univ., P.O. Box 206657, New Haven, CT 06520)

The goal of this experimental work was to ascertain to what degree the physical properties and oscillatory behavior of a material can be understood using only one's ability to hear and a few simple tools such as a ruler, a scale, and a piano. More specifically, by striking a metal pipe in various controlled ways, listening to the resonance frequencies of various modes, and then matching the pitches heard to those found on a piano, such physical properties as the Young's modulus and the velocity of sound in a medium (air and solid) can be derived. Similarly, more complex scenarios such as end corrections in air resonances and transverse modes of vibration can be described fairly accurately using only these prescribed tools. The accuracy of this method was checked using traditional instrumentation (microphone and spectrum analyzer). [Advisor, Professor Robert E. Apfel.]

**2aEDb3. Nonlinear scattering of crossed ultrasonic beams in the presence of turbulence: Experiments performed with pulses.** Rebecca A. Manry and Murray S. Korman (Dept. of Phys., U. S. Naval Acad., Annapolis, MD 21402, korman@nadm.navy.mil)

The nonlinear scattering of two finite-amplitude mutually perpendicular crossed beams—interacting in the presence of turbulence—generates a sum frequency component that radiates outside the interaction region. Experiments are reported where two primary pulsed ( $f_1 = 2.35$  MHz and

$f_2 = 1.65$  MHz) focused beams are generated by 2.54-cm-diam concave spherical transducer units ( $T_1$  and  $T_2$ ) of focal length 14 cm. The 4-MHz receiving unit ( $R$ ) is a 2.54-cm-diam circular plane array. The turbulence is generated by a  $D = 0.635$ -cm-diam submerged circular water jet (nozzle exit velocity 7 m/s) whose orifice is located at  $56D$  from the interaction region. All transducer beam axes and jet axis form a common plane. A scattering region is formed at the intersection of the focal points of the primary beams. While  $T_1$  and  $T_2$  rotate on radius arms—always keeping the beams perpendicular— $R$  is fixed. Symmetry suggests a scattering angle  $\theta_*$ , where  $\theta_* = 0$  defines forward scattering. Ensemble averaged rms pressure spectra (near the sum frequency) are measured (1) as a function of angle and fixed pulse duration  $\tau = 100$   $\mu$ s, and (2) at  $\theta_* = 0$  and variable  $\tau$  in an effort to compare spectral broadening with our theory.

**2aEDb4. Objective measurements in Strauss Recital Hall at the University of Nebraska.** Jessica L. Rock and Lily M. Wang (Architectural Eng. Prog., Univ. of Nebraska—Lincoln, 200B PKI, Omaha, NE 68182-0681)

This poster presents measured acoustical data in a performing arts hall, obtained from three different measurement techniques and with various sound source and receiver positions. The room under study is Strauss Recital Hall, located on the campus of the University of Nebraska at Omaha. Sound decay measurements are taken between selected source and receiver positions in three ways: triggered by an impulse source such as a balloon pop; triggered by the end of steady-state white noise; and using a maximum length sequence (MLS) source. The measured responses are used to calculate the reverberation time, clarity index, strength, interaural cross-correlation coefficient, and other room acoustic measures. Analysis of the data focuses on how these objective measures change across the Hall and how they differ between measurement techniques. Additionally, the data obtained are compared to values from other renowned halls.

**2aEDb5. Nonlinear scattering of crossed ultrasonic beams in the presence of turbulence: Experiments performed with continuous waves.** Marlene Roush and Murray S. Korman (Dept. of Phys., U. S. Naval Acad., Annapolis, MD 21402, korman@nadm.navy.mil)

Experiments are reported where two primary cw ( $f_1 = 2.35$  MHz and  $f_2 = 1.65$  MHz) beams are generated by 2.54-cm-diam transducer units ( $T_1$  and  $T_2$ ) of focal length 14 cm. The 4-MHz receiving unit ( $R$ ) is a 2.54-cm-diam circular plane array. Turbulence is generated by a  $D = 0.635$ -cm-diam submerged water jet (exit velocity 7 m/s) whose orifice is  $56D$  from the interaction region. All transducer beam axes and jet axis form a common plane. The ensemble averaged spectrum (near 4 MHz) is measured (using a Tektronix digital oscilloscope) as a function of the

“symmetry” scattering angle  $\theta_*$ . After transferring the data to a computer, “notebooks” in Mathematica are developed to curve fit the Doppler shift  $\langle f_d \rangle$  and the  $n=2$ nd, 3rd, and 4th spectral moments  $\langle (f - \langle f_d \rangle)^n \rangle$  vs  $\theta_*$  to the theory in an effort to measure the mean flow and turbulent

velocity correlation functions of the turbulence [Korman and Parker, in *13th ISNA, Bergen 1993, Advances in Nonlinear Acoustics*, edited by H. Hobaek (World Scientific, Singapore, 1993), pp. 650–655]. For example, for  $n=2$ , 2nd order correlations  $\langle v_x^2 \rangle$ ,  $\langle v_x v_y \rangle$ , and  $\langle v_y^2 \rangle$  can be measured.

TUESDAY MORNING, 5 JUNE 2001

WABASH ROOM, 9:30 TO 11:45 A.M.

## Session 2aNS

### Noise, Architectural Acoustics and Psychological and Physiological Acoustics: Measurement Procedures in Social Studies on Soundscapes and Annoyance

Brigitte Schulte-Fortkamp, Chair

*Physics-Acoustics, Carl von Ossietzky University, Carl Von Ossietzky Street, Oldenburg D-2611, Germany*

Chair's Introduction—9:30

#### Invited Papers

9:35

**2aNS1. Neighborhood soundscapes: Measurement and identification issues.** Ronny Klåboe (Inst. of Transport Economics, P.O. Box 6110 Etterstad, N-0602 Oslo, Norway)

The term neighborhood soundscape has been coined to describe the amount of road traffic noise in the immediate neighborhood of a dwelling. A simple indicator of how much the road traffic noise level in the neighborhood exceeds the noise level at the dwelling has been developed using a geographical information system. By applying the indicator with data from several socio-acoustic surveys it has been shown that people living in a relatively noisy neighborhood are significantly and substantially more annoyed than follows from the residential road traffic noise level alone. However, this does not prove that the heightened annoyance is caused by the noise exposure in the neighborhood. Neighborhood road traffic also gives rise to traffic insecurity and local air pollution, and indicators of these environmental exposures are additional possible effect modifiers. To separate the different positively correlated modifying effects, it is therefore not enough to measure the neighborhood soundscape, but also measure relevant aspects of the enviroinscape. To statistically separate the different modifying effects requires that the respective measures are sufficiently precise. To go beyond the purely statistical associations it is also necessary to link each measure to the respective psychscape impacts and model the relevant parts of the psychscape.

10:00

**2aNS2. Impression of sound environments formed by verbal explanation.** Kouji Abe, Toshio Sone (Akita Prefectural Univ., 84-4 Ebinokuchi, Tsuchiya, Honjo 015-0055, Japan, koji@akita-pu.ac.jp), Yōiti Suzuki (Tohoku Univ., Sendai 980-8577, Japan), and Kenji Ozawa (Yamanashi Univ., Kofu 400-8511, Japan)

Our research results suggested that additional verbal and visual information on sound environments may recall existing impressions of specific sounds and affect the auditory evaluation of environmental sounds (Y. Suzuki *et al.*, Proc. Internoise 2000, 2000, pp. 2285–2291). In order to verify this result, we conducted a new series of experiments as follows: At first, only a written verbal explanation of sound environments on sound fields and sound sources was presented to new subjects. In this experiment, we expected that some impression on the sound environments could be formed by the verbal explanation. Then, an experiment in which the stimulus sound only was presented as well as another experiment in which the stimulus sound was presented along with the verbal explanation were also conducted. The results from these experiments were analyzed by factor analysis. The results showed that the factors to evaluate the sound environments based only on the verbal information were similar to those in the evaluation with listening to the sounds. Moreover, analysis of the significant changes of factor scores among experiments showed that the performed impression on the sound environments by the verbal information affects evaluation of the sound.

10:25

**2aNS3. How structural parameters values can act on subjects' preference judgments for sounds transmitted by a simple component of an exterior wall.** Cathy Marquis-Favre and Julien Faure (LASH/DGCB, URA CNRS 1652, ENTPE, rue M. Audin, 69518 Vaulx-en-Velin, France, cathy.marquis@entpe.fr)

For a long time, the vibroacoustic behavior of structures was studied from a physical point of view with the goal of reducing sound levels and understanding physical phenomena. It has led researchers to elaborate upon physical models for sound radiation and transmission prediction, and also to establish vibroacoustic indicators such as radiation efficiency or sound transmission loss for acoustic performance criteria. Within the last decade, qualitative aspects of sounds have been integrated into the physical analysis to take into account sound perception and the expectation of subjects. This has yielded noticeable improvements in the sound quality of sources in various industries, mainly in the automotive industry. Our work aims at improving the acoustic performance of building structures which act as input filters to annoying sounds of the inner environment. The approach is based on coupling of the physical

and perceptual analyses. We will present how structural parameter values and their accuracy can act in a non-negligible manner on the similarity and preference judgment of subjects for broadband noises transmitted by a simple panel of an exterior wall. This is observed even in the case of structural parameter variations that induce no significant modification of vibroacoustic indicators such as sound transmission loss.

10:50–11:05 Break

11:05

**2aNS4. Soundscapes in the sense of reaction to sound and vibration.** Brigitte Schulte-Fortkamp (Dept. of Phys./Acoust., Oldenburg Univ., 26111 Oldenburg, Germany, [brigitte@aku.physik.uni-oldenburg.de](mailto:brigitte@aku.physik.uni-oldenburg.de))

Evaluation of soundscapes in urban areas has to focus on different aspects like the structure of urban areas, people living in those areas, architectural and social parameters designing those areas, and acoustical and visual parameters. A measurement is needed which refers to objective and subjective parameters. The structure of the residential area, the combination of noise sources, and the soundscape are for the judgment of annoyance as well as important as subjective parameters which are relevant according to the people's point of view; moreover the relationship of both define the background for assessments. When carrying out a survey on noise and vibration different subject-centered methodological procedures have been taken into account to develop a suitable measurement procedure. The survey focusing on perception of sound and vibration was conducted with about 600 subjects from different European countries. Procedures and results of these tests will be presented with respect to improvement of social surveys, especially addressing the meaning of noise and vibration in a defined environment. [Work supported by BRITE-EURAM project IDEA PACI/BE97-4056.]

### Contributed Paper

11:30

**2aNS5. Correlation between acoustical sensory profiles and physical measures of product sound.** David L. Bowen and Richard H. Lyon (RH Lyon Corp, 691 Concord Ave., Cambridge, MA 02138)

“Sensory profiles” (SPs) are used extensively for judgments of taste, odor, and texture—situations where a useful physical metric is not available. An acoustical SP can be obtained using a panel of critical listeners, and can provide a bridge between descriptions of a sound and the sound quality (SQ)—judgments that consumers make about a product based on

the sound it makes. But as descriptors, SPs also offer the opportunity to connect to physical measures of the sound. If the description is “tinny” or “shrill,” there is expectation of a high-frequency bias to the sound. If the description is “throbbing” or “knocking,” a modulation of the sound is expected. Since spectral balance and modulation are measurable, some kind of transformation between the SP and physical measurements is reasonable. This presentation focuses on the issue of the robustness of such a relationship, since it is desired that the physical measurements be able to predict or relate to a significant variation in the SP. Examples are given from studies of vacuum cleaner and washing machine sounds of the sensitivity of SQ to changes in the SP, and therefore of the variations in physical measures that are significant in terms of variations in SQ. [Work supported by NSF.]

TUESDAY MORNING, 5 JUNE 2001

PARLOR A, 8:15 TO 9:45 A.M.

### Session 2aPAa

#### Physical Acoustics: Interfaces, Particles and Bubbles

Kerry W. Commander, Chair

*Coastal Systems Station, Code R21, Panama City, Florida 32407-7001*

#### Contributed Papers

8:15

**2aPAa1. Passive stabilization of capillary bridges in air with acoustic radiation pressure and the equilibrium shape of bridges.** Mark J. Marr-Lyon, David B. Thiessen, and Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

Long liquid cylinders ordinarily become unstable because of a capillary instability originally studied by Rayleigh and Plateau. In the present research liquid bridges in air were acoustically stabilized to 43% beyond the natural limiting length identified by Rayleigh and Plateau [Marr-Lyon, Thiessen, and Marston, *Phys. Rev. Lett.* (to be published)]. The stabiliza-

tion was achieved by placing the bridge at the velocity antinode of a standing wave and selecting the wavelength such that the surface-averaged radiation pressure of the sound field increases with increasing bridge radius. The tests were performed on NASA's KC-135 aircraft in weightless conditions on bridges having volumes equal to that of a circular cylinder of the same length as the bridge. In the standing wave, the bridge becomes elliptical because of the spatial distribution of the radiation pressure. Calculations of the scattering of sound by elliptical cylinders using Mathieu functions show that elliptical deformation only weakly affects the average increase in the radiation pressure with the local bulging of a liquid bridge normally associated with the Rayleigh-Plateau instability. [Work supported by NASA.]



**2aPAa2. Simulation of particle motions in an acoustophoresis device.**

Todd L. Brooks and Robert E. Apfel (Dept. of Mech. Eng., Yale Univ., New Haven, CT 06520-8286, todd.brooks@yale.edu)

Particles in a liquid host can be continuously fractionated by combining a laminar flow with an ultrasonic standing wave oriented perpendicular to the flow direction. This kind of separation method can separate particles based on their relative compressibilities as well as on mass density and size. Experimental work has focused on separating particles ranging in size from 5 to 55  $\mu\text{m}$  in resonant devices operating from 100 kHz to 1 MHz. A numerical simulation based on a hierarchical tree method has been developed to simulate particle motion in such a device. The flow field, primary acoustic forces, and secondary interparticle acoustic forces are all taken into account. Full three-dimensional simulations can be performed. This method may be useful in other application areas where large numbers of particles, drops, and/or bubbles are under the influence of acoustic forces. [Work supported by NASA through Grant No. NAG8-1351.]

**2aPAa3. Acoustic cavitation thresholds and stable oscillations of vapor-gas bubbles.** Nail A. Gumerov (Inst. for Adv. Computer Studies, Univ. of Maryland, College Park, MD 20742)

The effect of rectified diffusion on the behavior of gas bubbles and the effect of rectified heat transfer to vapor bubbles are well established. These effects may be equally important for dynamics of vapor-gas bubbles and control the acoustic cavitation thresholds. To describe these effects a model of a spherical vapor-gas bubble is used. This model accounts for heat and mass transfer in the two-component system, including Soret-Dufour effects, kinetics of evaporation, surface tension, liquid viscosity, and compressibility. Equations for acoustic cavitation thresholds and stable bubble oscillations are obtained using asymptotic averaging techniques valid to second order in the amplitude of the acoustic field. Good agreement between the previous results for gas and vapor cavitation thresholds and the present computations is observed in the limiting cases of single-component bubbles. The effects of temperature, kinetics of evaporation, supersaturation, and others on the acoustic cavitation thresholds and stable bubble oscillations are studied and discussed. The existence of stable oscillations and cavitation thresholds of the bubble due to the second resonance of a vapor-gas bubble is predicted. These states occur due to the two-component nature of the bubble and appear at low inert gas concentrations near the liquid boiling point.

**2aPAa4. Single bubble sonochemistry and sonoluminescence.** Yuri T. Didenko and Kenneth S. Suslick (Dept. of Chemistry, Univ. of Illinois–Urbana-Champaign, 601 S. Goodwin Ave., Urbana, IL 61801)

An isolated, single vapor-gas bubble pulsating in a standing acoustic field can emit flashes of light. This peculiar phenomenon is known as single bubble sonoluminescence, SBSL. We find that these bubble pulsations in water are accompanied with the formation of radicals and molecular products. In this paper, the yields of hydroxyl radicals and nitrite ions formed inside the bubble were measured for the first time. The chemical

and light efficiency of acoustic cavitation and the diffusion rate of nitrogen inside the bubble during its pulsation were calculated using experimental data. The energy efficiency of OH radical formation by a single bubble is comparable to that in multibubble cavitation. However, the energy efficiency of light emission is much higher for SBSL. The diffusion rate of nitrogen inside the bubble is in good agreement with that predicted by the dissociation hypothesis. [Work supported by NAVY/ DARPA.]

**2aPAa5. Towards an intense bubble.** Anthony Khong, Elizabeth Doschek, Jonathan Mori, Robert Apfel (Dept. of Mech. Eng., Yale Univ., 9 Hillhouse Ave., New Haven, CT 06520, anthony.khong@yale.edu), and Jeffrey Ketterling (Riverside Res. Inst., New York, NY 10036)

A main goal in SBSL remains the ability to form a bright and hot bubble. One recent postulate suggests that lowering the resonance frequency should favor a stronger luminescing bubble. It is also theorized that the chemical composition of the bubble can be streamlined to induce a more violent collapse. The water vapor content inside the bubble is considered. The premise of this conjecture is that water vapor molecules, present beyond equilibrium concentrations, act to siphon energy from the bubble, and in the process undergo thermal decomposition. Hence, reducing the water vapor content should eliminate such energy loss. The current study focuses on this issue. Keeping similar driving conditions, there is interest in using aqueous salts to depress the vapor pressure of the water in tandem with a reduction in the water temperature. A series of alkali metal halides will be examined. In a preliminary investigation, a 9% lithium chloride solution saturated with 1% argon gave stable SL at room conditions. With air, no SL was detected. A further extension of this work would be to determine the bubble temperature which has largely been inferred. [Work supported by DARPA through a subcontract from the University of Washington.]

**2aPAa6. Focused pulse of pressure for boosting sonoluminescence.**

Jean-Louis Thomas, Yoel Forterre, and Mathias Fink (Lab. Ondes et Acoustique, CNRS UMR 7587, Univ. Paris VII, ESPCI, 10, rue Vauquelin 75231 Paris, France)

Single bubble sonoluminescence is characterized by a great concentration of energy conducting to the generation of UV photon with ultrasound of centimetric wavelength. Several teams have tried to again increase more of the energy concentration, but most of these trials have rapidly reached the boundaries of the narrow stability domain of sonoluminescence. The idea developed in this work is to overcome this limitation by adding a high-frequency pressure pulse to the monochromatic low-frequency field used in the now classical setup. For instance, the pressure in the neighborhood of the bubble can be dramatically increased during a single collapse of the bubble. This pressure pulse is generated with eight high-frequency transducers adaptively focused on the bubble and synchronized with its low-frequency cycle. With this setup, a transient amplitude of 8 atmospheres has been reached on the bubble, without destroying it, and a significant increase of the emitted light intensity has been observed on a single flash. This process is no longer limited by the domain of stability of sonoluminescence and could lead, in the future, to much more efficient inertial confinement of the gas contained in the bubble.

## Session 2aPAb

## Physical Acoustics: Sensors and Devices

Robin O. Cleveland, Chair

Department of Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street,  
Boston, Massachusetts 02215

## Contributed Papers

10:00

**2aPAb1. Pulse-echo ultrasonics from the correlations of thermal phonons.** Richard L. Weaver and Oleg I. Lobkis (Theoretical and Appl. Mech., Univ. of Illinois, Urbana, IL)

A high-sensitivity acoustic emission transducer in contact with a non-insonified solid body generates a noise signal, a significant fraction of which is due to thermal fluctuations in the body. This is established by comparisons of the spectral power density of its noise and the spectral power density consequent to the application of a calibrated source. This high sensitivity is some 30 dB greater than that of other ultrasonic transducers. As established elsewhere [Weaver and Lobkis, this meeting] the correlations of a diffuse field are related to the local density-of-ultrasonic-modes, and thus to local responses in pulse/echo. It is therefore hypothesized that the autocorrelation function of the noise in a high-sensitivity transducer should equal the pulse-echo signal generated by that transducer. The hypothesis is investigated by constructing that autocorrelation function (the autocorrelation appears to converge after only a few tens of milliseconds of data collection) and comparing it with the pulse/echo signal. The two waveforms are found to be substantially identical. The reasons for the differences are discussed, and possibilities for application are considered. [Work supported by NSF.]

10:15

**2aPAb2. Diffraction correction and attenuation of high-frequency acoustic pulses in a relaxing medium.** Yefim Dain and Richard M. Lueptow (Dept. of Mech. Eng., Northwestern Univ., Evanston, IL 60208, rlueptow@northwestern.edu)

The influence of intrinsic absorption in a relaxing medium and the diffraction correction for the magnitude of the acoustic pressure averaged over the surface of a receiver was investigated. An exact formula for the damped acoustic pressure averaged over the surface of the receiver was obtained for high-frequency modulated pulses in a mono-relaxing medium. The formulation is based on the solution of a modified wave equation in a relaxing medium. Based on asymptotic analysis, separate effects result for the attenuation of the pulse envelope and for the modulation of the carrier frequency for high-frequency pulses with an arbitrary envelope shape. The resulting distortion of the energy spectrum for arbitrary pulses depends on the medium's relaxation frequency and the acoustic carrier frequency. These results are useful in predicting the response of transducers in a relaxing medium. [Work supported by Ford Motor Company.]

10:30

**2aPAb3. Cuts with a negative Poisson's ratio in Si.** Svetlana Tokmakova (N. Andreyev Acoust. Inst., Shvernika Str. 4, Moscow 117036, Russia)

Poisson's ratio is the ratio of lateral strain in the direction  $n$  to the longitudinal strain during the stretching of a cylindrical rod along rod axis  $m$ . The Poisson ratio is a constant in an isotropic solid. The Poisson ratio of an anisotropic solid depends on directions  $m$  and  $n$ . In the paper the stereographic projections of Poisson's ratio for some crystals were com-

puted (germanium, quartz, lithium niobate, langasite, zinc, copper, etc.). From the stereographic projections the Poisson ratio for any directions of  $m$  and  $n$  were calculated. Orientations of axes of a rod  $m$  and directions of lateral strain  $n$  with extreme (negative) values of Poisson's ratio were determined. Only for Si and HgCl was a decrease of rod cross-section under the uniaxial compression predicted for some orientations of the rod axis. For the Si rod the axis of the rod should lay within the 30-deg cone (the axis of the cone is the [001] direction of the Si crystal). Materials having a negative Poisson ratio may find a lot of applications, for example, as electrodes that amplify the response of piezoelectric sensors [Baughman *et al.*, Nature (London) **392**, 362–365 (1998)].

10:45

**2aPAb4. A nonlinear analysis of piezoelectric coated laminae.** M. Cengiz Dokmeci (Istanbul Tech. Univ., Gumussuyu, P.K.9, Istanbul 80191, Turkey) and G. Askar Altay (Bogazici Univ., Bebek, Istanbul 80815, Turkey)

In relation to vibrations of piezoelectric materials subject to strong electric fields and large strains, a system of 2-D nonlinear laminae equations is derived by the use of Mindlin's kinematic hypothesis for each layer. The laminae is coated completely with perfectly conducting electrodes on both its faces, and it may comprise any number of bonded layers, each with a distinct but uniform thickness and electromechanical properties. By use of a recent variational principle [Altay and Dokmeci, Thin Walled Struct. **39**, 95–109 (2001)], the system of equations is consistently formulated in both invariant, differential and variational forms. The effects of elastic stiffnesses of, and the interactions between, layers of laminae are all taken into account. The system of equations governs the extensional, flexural, and thickness shear as well as coupled vibrations of piezoelectric laminae. Special types of vibrations, geometry, and material properties are recorded. Besides, the uniqueness is investigated in solutions of the initial-mixed boundary value problems defined by the linearized system of laminae equations. [Work supported in part by TUBA.]

11:00

**2aPAb5. Crosstalk between acoustic sensors in a MEMS array.** Allan D. Pierce, Harley T. Johnson, Robin O. Cleveland, Daniel N. Pascual, and Thomas G. Bifano (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215)

The sensor array considered is in bare essentials a regular rectangular array of square holes in a slab of single crystalline silicon. The actual sensing of an incident pressure wave is achieved by optical sensing of the acoustically induced vibrations of recessed membranes (plates) within the holes. The present study addresses the crosstalk question by investigating the effect of the spacing between the holes (sensors) on the vibration of the individual membranes. The no crosstalk limit is taken as when adjacent holes are asymptotically very far apart. When they are closer together the amplitude changes because of two primary effects: (1) elastic waves originating at each of the individual holes and stimulating vibrations of the membranes in adjacent holes, and (2) reradiation of sound from one hole and subsequent diffraction into an adjacent hole. Complementary analyses

are carried out using a finite element program and the geometrical theory of diffraction. For the array under study the dominant effect results from diffraction through the fluid itself; structure-borne sound has a minor effect.

11:15

**2aPAb6. Microfluidic amplification of ultrasound with application to MEMS hydrophones.** Robin O. Cleveland, Daniel N. Pascual, Harley T. Johnson, Thomas G. Bifano, and Allan D. Pierce (Dept. of Aerosp. and Mech. Eng., Boston Univ., Boston, MA 02215)

Micro-electro-mechanical systems (MEMS) were used to fabricate a device with passive fluidic components that amplified the motion associated with ultrasonic waves in water. Two types of fluid components were etched in silicon, an acoustic horn, and an acoustic resonator. The fluid motion was detected by a small silicon diaphragm which moved with the sound wave. The diaphragm motion was measured with a laser Doppler vibrometer. Experiments were carried out for acoustic excitations in the frequency region of 0.5 to 5 MHz. Over this frequency range uniform thickness diaphragms were found to support multiple resonance modes and were found to be poor sensing surfaces. Composite diaphragms, with a stiff pedestal supported by compliant edge fixtures, provided a superior sensing surface. The motion of the diaphragm placed at the throat of an

acoustic horn demonstrated broadband amplification with a gain of 6. An organ-pipe type resonator provided a narrow-band gain of 3. The horn and resonator structures could be incorporated into most types of MEMS hydrophone devices which use diaphragms to sense sound waves. [Work supported by DARPA.]

11:30

**2aPAb7. The scaling of acoustic streaming for application in microfluidic devices.** Kenneth Frampton (Vanderbilt Univ., Box 1592 B, Nashville, TN 37235)

Recent interest in MEMS devices in general, and in microfluidic devices specifically, has spurred a great deal of research into the behavior of fluids in very small-scale devices. Many novel techniques have been proposed for the propulsion of fluids in small-scale devices including peristaltic and electrokinetic. More recently, investigations have considered the use of acoustic streaming, that is, the imposition of steady fluid momentum via nonlinear acoustic effects. The purpose of this talk is to give an overview of the physics of acoustic streaming, to discuss the various physical phenomena which generate the effect, and to highlight the favorable scaling issues of acoustic streaming that make it a viable alternative in microfluidic devices.

TUESDAY MORNING, 5 JUNE 2001

GRAND BALLROOM, 8:45 TO 10:45 A.M.

## Session 2aPPa

### Psychological and Physiological Acoustics: Masking and Loudness

Mary Florentine, Chair

*Department of Speech Language Pathology and Audiology, Northeastern University, (133FR), 70 Forsyth Street, Boston, Massachusetts 02115*

#### Contributed Papers

8:45

**2aPPa1. Effect of masker harmonicity on threshold: Auditory filter envelope-modulation rate variability versus equidistant spacing of partial frequencies.** William C. Treurniet and Darcy R. Boucher (Commun. Res. Ctr., Box 11490, Sta. H, Ottawa, ON K2H 8S2, Canada, bill.treurniet@crc.ca)

In a previous paper [W. C. Treurniet and D. R. Boucher, J. Acoust. Soc. Am. **109**, 306–320 (2001)], a model was presented which successfully accounted for the lower threshold obtained using a harmonic masker instead of a similar inharmonic masker. For the harmonic masker, the frequencies of partials were separated by a fixed amount, so the envelope modulation-rates of auditory filter outputs remained the same across filters. However, for the inharmonic masker, the interval between adjacent partials was not fixed and this resulted in a decreased uniformity of modulation-rates across filters. The model proposed that the lower uniformity impedes detection of a probe-induced change in the modulation rates, thus accounting for the masked threshold difference. This paper shows that an inharmonic masker yields results similar to a harmonic masker provided that the modulation rates are uniform across affected auditory filters. Thus, the lowered threshold associated with a harmonic masker appears to arise from invariant modulation-rates across auditory filters, and does not require that all partial-frequencies be integer multiples of a fundamental.

9:00

**2aPPa2. Upward spread of Schroeder-phase maskers.** Jennifer Lentz, Marjorie Leek (Army Aud. & Speech Ctr., Walter Reed Army Medical Ctr., 6900 Georgia Ave., NW, Washington, DC 20307, jennifer.lentz@na.amedd.army.mil), and Laura Dreisbach (San Diego State Univ., San Diego, CA 92182-1518)

Harmonic complexes with phases selected according to the Schroeder algorithm can produce large differences in masking, depending on whether phases increase (positive Schroeder) or decrease (negative Schroeder) with frequency. This finding has been attributed to an interaction between the phase characteristics of auditory filters and the stimulus. The current study investigates effects of filter asymmetry on masking by harmonic complexes using an upward spread of masking paradigm. Maskers were Schroeder phase maskers with frequencies from 200 to 2000 Hz, and signal frequencies ranged from 1000 to 4000 Hz. When the signal fell within the masker bandwidth, it was added in-phase with the identical masker component. At signals below 2000 Hz, negative Schroeder maskers produced more masking than the positive maskers. When the signal frequency was above the masker, there was a rapid decrease in masking for both maskers, but for most subjects, the positive masker became more effective than the negative. This shift in the Schroeder masking effect may be related to phase changes occurring in the lower skirts of the auditory filters. [Work supported by NIH.]



**2aPPa3. Relation between the slope of the rate-level curve, the maximum driven firing-rate, and the neuronal dynamic range in auditory primary afferents of the cat.** Lance Nizami, JoAnn McGee, and Edward J. Walsh (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, nizami@boystown.org)

The power of the auditory primary afferent to indicate intensity change depends partly upon the rate-of-change-of-firing with sound-pressure-level, i.e., the slope of the rate-level curve. The latter was estimated for each of 778 rate-level-functions in the cat (McGee, MS thesis, Creighton, 1983) by fitting lines to the points on the curve that fall between (1)  $R_s + [c/100](R_{\max} - R_s)$  and (2)  $R_s + [(100-c)/100](R_{\max} - R_s)$ , for  $c = 30$ .  $R_s$  is spontaneous firing-rate and  $R_{\max}$  is maximum firing-rate.  $c$  specifies a pair of points on the curve, symmetric around the centroid. Each slope was divided by the respective maximum driven firing-rate,  $R_{\max} - R_s$ , to obtain the adjusted slope (in 1/dB). Each adjusted slope was then plotted versus the neuron's dynamic range (in dB), whose endpoints, the threshold and the saturation level, are given by  $c=1$ . The resulting plot is smooth and hyperbolic. Threshold and saturation level have also been defined by expressions (1) and (2) in a logistic rate-level equation (Nizami, 24th ARO, 2001) that yields (3) adjusted slope  $= [\ln[(100-c)/c]]/2(\text{dynamic range})$ . For  $c=1$  and  $c=2$ , these curves neatly sandwich the empirical data. An afferent's power of discrimination is intimately related to both the maximum driven firing-rate and the neuronal dynamic range.

9:30

**2aPPa4. Derivation of intensity-discrimination functions from loudness functions.** William S. Hellman (Dept. of Phys. and Hearing Res. Ctr., Boston Univ., 590 Commonwealth Ave., Boston, MA 02215) and Rhona P. Hellman (Northeastern Univ., Boston, MA 02115)

Intensity-discrimination functions are computed from their concomitant loudness functions in a procedure where the only adjustable parameter sets the scale to match the measured intensity-discrimination data. The equation connecting loudness to intensity discrimination is determined from monaural loudness- and intensity-discrimination measurements at 1 kHz. Predictions of intensity-discrimination functions are then computed for a low-frequency tone masked by an adjacent high-pass noise, for broadband noise, and for high-frequency tones at 12.5 and 16 kHz. Results show that the derived intensity-discrimination functions closely capture the overall shape of the experimental data. A U-shaped function is obtained for the partially masked low-frequency tone, whereas a function approximating Weber's law is observed for broadband noise and also at 16 kHz. The good agreement between the calculated and measured functions implies that the form of the intensity-discrimination functions is strongly dependent on local variations in shape of the loudness functions. Together, these results provide further evidence to support the notion of an intimate relation between loudness and intensity discrimination. [Work supported in part by NEDO, Japan.]

9:45

**2aPPa5. Loudness functions for long and short tones.** Mary Florentine (Inst. for Hearing, Speech and Lang. and Dept. of Speech-Lang. Path. and Audiol., 133 FR, Northeastern Univ., Boston, MA 02115, florentin@neu.edu), Michael Epstein, and Søren Buus (Inst. for Hearing, Speech and Lang., Northeastern Univ., Boston, MA 02115)

This study tests the equal-loudness ratio hypothesis [Florentine *et al.*, J. Acoust. Soc. Am. **99**, 1633–1644 (1996)], which states that the loudness ratio between equal-SPL long and short tones is independent of SPL. The amount of temporal integration (i.e., the level difference between equally loud long and short sounds) is maximal at moderate levels. Therefore, the equal-loudness ratio hypothesis predicts that the loudness function is shallower at moderate levels than at low and high levels. Equal-loudness matches and cross-modality string-length matches were used to assess the form of the loudness function for 200- and 5-ms tones at 1 kHz. Results from nine normal listeners show that (1) the amount of temporal integration is largest at moderate levels in agreement with previous studies, and (2) the loudness function is shallowest at moderate levels. For eight of the

nine listeners, the loudness ratio between the 200- and 5-ms tones is approximately constant, except at low levels where it tends to increase. The average data show good agreement between the two methods, but discrepancies are apparent for some individuals. These findings support the equal-loudness ratio hypothesis, except perhaps at low levels. [Work supported by NIH/NIDCD Grant No. R01DC02241.]

10:00

**2aPPa6. Equal-loudness contours at high frequencies reconsidered.** Rhona Hellman (Dept. of Psych., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115), Hisashi Takeshima (Sendai Natl. College of Technol., Sendai 989-3124, Japan), Yôiti Suzuki (Tohoku Univ., Sendai 980-8577, Japan), Kenji Ozawa (Yamanashi Univ., Kofu 400-8511, Japan), and Toshio Sone (Akita Prefectural Univ., Honjo City 015-0055, Japan)

To add to the database and to clarify the spacing of the equal-loudness contours at high frequencies, equal-loudness relations determined from 1 to 16 kHz in a recent study [Hellman *et al.*, J. Acoust. Soc. Am. **103**, 2812 (1998)] are reevaluated. Relative to the linear loudness-level function with a slope of 1.0 observed for a standard 1-kHz tone, above 10 kHz the overall shapes and slopes of the loudness-level functions are both level and frequency dependent. Below 60 phons, the slopes of the loudness-level functions at 12.5 kHz and higher increase progressively with frequency from 1.31 at 12.5 kHz to 1.79 at 16 kHz. Conversely, above 60 phons the slopes decrease from 0.98 at 12.5 kHz to 0.74 at 16 kHz. The data imply that for frequencies between 1 and 10 kHz, the spacing between the equal-loudness contours is independent of loudness level. In contrast, above 10 kHz the equal-loudness contours are more closely spaced below 60 phons than at higher loudness levels. Moreover, in accord with the early work of Robinson [IRE Trans. Audio **1**, 6–13 (1958)], the higher the frequency, the more strongly does the spacing vary with loudness level. [Work supported by NEDO, Japan.]

10:15

**2aPPa7. Effect of duration and level of the recalibration tone on the amount of loudness recalibration.** Bärbel Nieder (Inst. of Hearing, Speech, and Lang. and Dept. of Speech-Lang. Path. and Audiol., 133 FR, Northeastern Univ., Boston, MA 02115, nieder@research.neu.edu), Søren Buus (Inst. of Hearing, Speech, and Lang. and Commun. and Dig. Sig. Proc. Ctr., Northeastern Univ., Boston, MA 02115), Mary Florentine (Inst. of Hearing, Speech, and Lang., Northeastern Univ., Boston, MA 02115), and Bertram Scharf (Inst. of Hearing, Speech, and Lang., Northeastern Univ., Boston, MA 02115)

The amount of loudness recalibration (i.e., the drop in loudness of a moderate-level tone caused by a preceding intense recalibration tone of the same frequency) was measured as a function of the recalibration tone's duration and level using a 2AFC procedure. The 500-Hz test tone—presented at 60 and 70 dB SPL—and the 2500-Hz variable-level comparison tone both lasted 200 ms. Results for 10 normal listeners show that 5-ms recalibration tones yielded only 3 (test tone at 60 dB SPL) to 4 dB (at 70 dB SPL) of recalibration, whether their level was 80, 95, or 110 dB SPL. In contrast, 200- and 500-ms recalibration tones at 80 and 95 dB SPL (200 ms only) yielded 6 (at 60 dB SPL) to 10 dB (at 70 dB SPL) of recalibration, again with no apparent effect of level. Note that 5-ms recalibration tones at 95 dB SPL yielded much less recalibration than 200-ms recalibration tones at 80 dB SPL, despite their nearly equal loudness. These data indicate that recalibration is not governed by the loudness of the recalibration tone and that recalibration-tone duration is a crucial parameter for recalibration of loudness. [Supported by NIH/NIDCD R01DC02241.]

**2aPPa8. Further progress with loudness adaptation inside and outside the speech range.** Sophia Boudouris, Kathleen Cross, Suzanne Boyce, Laura Kretschmer, David E. Sandman, and Ernest M. Weiler (ML #394, Psycho-Acoust. Lab., CSD, CAHS, Univ. of Cincinnati, Cincinnati, OH 45267-0394, Ernest.Weiler@uc.edu)

Measurement of loudness adaptation at 4000 Hz and below depends on the technique used [Weiler *et al.*, J. Acoust. Soc. Am. **101**, 3171(A)

(1997); T. Maguire *et al.*, J. Acoust. Soc. Am. **106**, 2207 (1999)]. Further comparison of simple adaptation (SA) to the ipsilateral comparison paradigm (ICP) with repeated measures designs again shows adaptation for both techniques at 8000 Hz but stronger effects for the ICP now. At 6000 Hz only, violations of normal curve parameters were again observed but no SA. Observations at 250 Hz for the ICP and SA will be discussed. The difference in adaptation between the two techniques is extreme in the primary speech frequencies but intermediate at 8000 Hz for our repeated measures designs. Further investigation is planned.

TUESDAY MORNING, 5 JUNE 2001

GRAND BALLROOM, 11:00 A.M. TO 12:05 P.M.

### Session 2aPPb

## Psychological and Physiological Acoustics and Archives and History: History of Physiological Acoustics in ASA

Peter Dallos, Chair

*Frances Searle Building, Northwestern University, 2299 Sheridan Road, Evanston, Illinois 60208*

Chair's Introduction—11:00

### Invited Paper

11:05

**2aPPb1. History of physiological acoustics: A modeling perspective.** Murray B. Sachs (Dept. of Biomed. Eng., Johns Hopkins Univ., 720 Rutland Ave., Baltimore, MD 21205)

Since the days of Helmholtz and Ohm in the nineteenth century, physiological acoustics has focused on models of mechanisms underlying human auditory perception. We review the history of three modeling streams: (1) models of cochlear mechanisms, (2) models that predict human psychoacoustic performance on the basis of neural representations of sound, and (3) models for processing of acoustic information by the central nervous system. We trace the evolution from early macromechanical models that were adequate to explain the broad tuning of the basilar membrane as reported by von Békésy to recent micromechanical models needed to account for sharp basilar-membrane tuning as revealed by more sensitive experimental techniques. We review models of the molecular mechanisms underlying hair cell electromotility and its relation to sharp tuning. Because of the "dynamic range problem" early models of frequency and intensity discrimination based on auditory-nerve discharge rates gave way to models based on temporal properties of auditory-nerve firing patterns. Discovery of a population of fibers with broad dynamic ranges revived interest in rate-based models. Studies of information processing in the central auditory system have been shaped by models for the representation of the frequency, intensity, and spatial characteristics of acoustic stimuli.

#### A NOTE ABOUT THE ASA HISTORY LECTURE SERIES

In 1997, the ASA Committee on Archives and History conceived a plan for a series of invited lectures on each of the technical areas of the Society which would memorialize the significant achievements and milestones of each of its technical committees during the first three quarters of the Society's first century.

With the cooperation of the technical committees, distinguished individuals are selected to review the history of their particular technical specialty and present a lecture which shows how that activity has developed and has contributed to the Society at large and to the broad field of acoustics as well.

The invited lecturers have been asked to prepare a written manuscript of their lectures which will be published in a commemorative book for the 75th Anniversary of the Society to be celebrated in 2004. The Archives and History Committee and the individual technical committees/group welcome comments and suggestions on both the History Lecture Series and on the proposed ASA Diamond Anniversary Book. Volunteers to assist the committees would be most welcome too. Contact Henry Bass, Chair, Committee on Archives and History, [pabass@sunset.backbone.olemiss.edu](mailto:pabass@sunset.backbone.olemiss.edu)

## Session 2aSAa

## Structural Acoustics and Vibration: Vibroacoustics and Noise Control

Kenneth A. Cunefare, Chair

*School of Mechanical Engineering, Georgia Institute of Technology, Graduate Box 268, Atlanta, Georgia 30332-0405*

## Contributed Papers

8:00

**2aSAa1. Theoretical and experimental modal analysis of brake pads.**

Mario Triches, Jr., Samir N. Y. Gerges (Mech. Eng. Dept., Federal Univ. of Santa Catarina, Cx.P. 476-Florianopolis-SC, 88040-900, Brazil), and Shahram Tousi (MSC Laminates and Composites, Inc., Elk Grove Village, IL 6007-5995)

Disc brake squeal noise is a complicated dynamic problem that has confronted automobile manufacturers for decades. The reduction of brake squeal noise is an important technological subject in terms of making vehicles quieter. Two main mechanisms are correlated with squeal noise in disc brake systems. The first one is related to the velocity dependency of the friction coefficient. The second one is the modal coupling of the rotor and the pads. In this case, the modes of the pads and rotor are coupled and the system may become unstable. To extract the natural frequencies and the vibration modes of pads is of great importance for the forecast of the dynamic behavior of the complete system. Thus, it can predict the behavior of the pads with the noise generated by the system. In this work, results of experimental modal analysis of the pads are presented and compared with the results obtained through use of the finite-element method (FEM). Different conditions of temperature were applied to the pads and simulated by FEM, verifying the influence of the work temperature in the natural frequencies.

8:15

**2aSAa2. Transfer matrix method for mufflers modeling and experimental verification.** Fabio A. Thieme (Eberspcher Tuper Sistema de Exhausto Ltda., So Bento de Sul, SC, Brazil), Samir N. Y. Gerges (Federal Univ. of Santa Catarina, Florianopolis, SC, Brazil), and Jos L. B. Coelho (Insituto Superior Tecnico, 1049-001 Lisboa, Portugal)

Mufflers are very important elements in industry for the attenuation of exhaust noise in vehicles and machinery. Recent advances in modeling procedures, for accurate performance prediction, lead to the development of the transfer matrices method for more practical muffler components appearing in commercial designs. Muffler designers are looking for simple, quick modeling tools, especially in the preliminary evaluation stages of a design. Finite element (FEM) and boundary element (BEM) methods, in spite of giving results in a wide range of frequencies, are still expensive, need very skilled operators, and commercial software is still not user friendly. Plane wave modeling, using the transfer matrix form, offers a cheap and quick method for muffler designers to arrive at an initial prototype solution. In this paper, transmission loss calculations are presented for several practical muffler configurations in explicit formulas and comparison with experimental results are presented.

8:30

**2aSAa3. Theoretical development and experimental verification of polyvinylidene fluoride sensors for measurement of the local volume displacement of beams.** Randall Rozema, Brian Zellers, Koorosh Naghshineh (Mech. and Aeronautical Eng. Dept., Western Michigan Univ., Kalamazoo, MI 49008), and Marcellin Zahui (Lake Superior State Univ., Sault Sainte Marie, MI 49783)

One method of reducing the sound radiated from vibrating structures at lower frequencies is to reduce the volume displacement of the vibrating surface (via active control). At these low frequencies, the volume displacement is directly proportional to the sound power emitted from a vibrating

surface. To extend the effective frequency range of the active control system, several systems that reduce the volume displacement over localized areas of structural surface were employed. Thus, means of measurement of structural surface volume displacement become important. A traditional approach is to employ multiple point sensors (accelerometers). Recently, a single sensor made of polyvinylidene fluoride (PVDF) was utilized and found to represent an attractive solution. These sensors, which were designed to measure the structural volume displacement over a segment of structural surface, spanned the entire length of the structure. Such arrangement was found to be inefficient. In the work presented here an integrated sensor comprised of a shaped PVDF strip in addition to two point sensors located at the ends of this strip. The entire sensor does not extend beyond the segment of interest, thus, labeled a local sensor. Theoretical development of these sensors, fabrication techniques, and verification of these sensors will be presented.

8:45

**2aSAa4. Investigation of local volume displacement sensors for rectangular vibrating plates.** Mark Christensen, Ron Hoofnagle, Koorosh Naghshineh (Mech. and Aeronautical Eng. Dept., Western Michigan Univ., Kalamazoo, MI 49008), and Marcellin Zahui (Lake Superior State Univ., Sault Ste. Marie, MI 49783)

Further development and validation of a technique for measurement of local volume displacement is presented. This development supports the implementation of noise control techniques that are based on minimization of local volume displacements, velocities, or accelerations of a vibrating structure. In this work, we present a brief description of the methodology for designing such sensors fabricated using polyvinylidene fluoride (PVDF) film followed by the experimental verification of these sensors for a vibrating clamped-clamped plate comprising one side of an otherwise rigid box. These experimental results were then compared against predicted values.

9:00

**2aSAa5. Experimental validation of the state-switched absorber for two-component harmonic forcing.** Mark Holdhusen, Kenneth A. Cunefare, and Gregg Larson (The George W. Woodruff School of Mech. Eng., The Georgia Inst. of Technol., Atlanta, GA 30332-0405, gte165r@prism.gatech.edu)

A state-switched vibration absorber (SSA) is a semiactive device that is capable of instantaneously changing its dynamical state by altering its stiffness. However, the SSA can switch between more than one resonance frequency, providing a greater bandwidth as compared to classical tuned vibration absorbers. Modeling predictions indicate that with appropriate logic for switching between states, the SSA is more effective at vibration suppression than classical tuned vibration absorbers, when the system to be controlled is subject to a disturbance with more than one spectral component. This presentation considers the experimental validation of SSA performance for the suppression of vibration on a driven elastically mounted lumped mass system. The SSA considered here is implemented in a dynamical proof-of-concept demonstrator. State switching is effected

2a TUE. AM

through the use of a magneto-rheological fluid clamp to connect or disconnect a coil spring in parallel with another coil spring. The two coil springs, one switchable, the other not, constitute the spring element of an otherwise passive mass-spring-damper vibration absorber. The stiffness state is controlled by actuation of the MR clamp. The experimentally observed performance of the SSA will be presented and compared to modeling predictions. [This work is based upon work supported by, or in part by, the U.S. Army Research Laboratory and the U.S. Army Research Office under Grant No. 38955-EG, Dr. Gary Anderson, Technical Monitor.]

9:15

**2aSAa6. A passive means for cancellation of structurally radiated tones.** Jeffrey A. Zapfe and Eric E. Ungar (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138-1118)

The mass of a spring-mass system whose spring is driven at a given frequency vibrates in opposite phase to the excitation, provided that the system's natural frequency is lower than the excitation frequency. If this natural frequency also is near the excitation frequency, then the amplitude of the mass can be considerably greater than that of the driving motion. These ideas were applied to construct radiation-canceling devices that act like resiliently supported masses with significant sound radiating areas, so that radiation from these could cancel the sound radiated from considerably larger vibrating surfaces to which these devices may be attached. The sound radiation from such devices and the attendant reduction in the sound radiated from sample structures were analyzed, some experimental proto-

type devices were built, and their effects were measured. Devices of this type are thought to be useful for reducing the sound radiation from equipment items, such as transformers, which vibrate and radiate noise at a fixed frequency. [Patent pending.]

9:30

**2aSAa7. Sound insulation analysis of solid walls using the finite element method.** Lutz Ackermann and Heinz Antes (Inst. of Appl. Mech., Tech. Univ. of Braunschweig, Spielmannstr. 11, 38106 Braunschweig, Germany, lackermann@tu-bs.de)

In times of increasing noise pollution, the numerical simulation of sound transmission through solid walls, e.g., of masonry, is a challenging building acoustics topic. For an adequate building design tool it is of great importance to take arbitrary geometrical and acoustical boundary conditions as well as the air-structure interaction into account. Here, a model based on the finite element method is presented for the calculation of sound propagation in layered air/wall/fiber systems. The structure-borne sound is, on the one hand, influenced by the bending waves and, on the other hand, by the in-plane waves, if also the flanking transmission is investigated, where the bending waves are modeled by the Mindlin plate theory and the in-plane waves by the dynamic elastic disk equation. The acoustic behavior of air is described by the Helmholtz equation. The fluid-structure interaction is performed by using the principle of virtual work, and, as a final result, a completely coupled methodology is derived which allows the determination of sound fields and of the transmission loss of conventional solid walls. Numerical results are compared with measured values and the influence of various parameters is studied.

TUESDAY MORNING, 5 JUNE 2001

CRYSTAL ROOM, 10:00 A.M. TO 12:00 NOON

## Session 2aSAb

### Structural Acoustics and Vibration: Analytical and Numerical Methods in Vibrations

Linda P. Franzoni, Chair

*Department of Mechanical Engineering and Material Science, Duke University, Box 90300, Durham, North Carolina 27708-0300*

#### Contributed Papers

10:00

**2aSAb1. Wave vibration analysis and control of a transversely vibrating Timoshenko beam.** Carole Mei (Dept. of Mech. Eng., The Univ. of Michigan-Dearborn, 4901 Evergreen Rd., Dearborn, MI 48128)

Apart from the modal description, vibrations in structures can also be described in terms of waves. In this paper, wave vibration analysis and control of a transversely vibrating Timoshenko beam are studied. The wave reflection and transmission characteristics under general discontinuity and general boundary conditions are presented. The reflection and transmission matrices for incident waves upon such discontinuities and boundaries are derived. Both free and forced responses of the Timoshenko beam are analyzed from wave reflection and transmission standpoint. The Timoshenko and the Euler-Bernoulli beam model are compared. Numerical examples of the free and forced vibrations of both slender and non-slender beams are presented to show the similarities and to signify the differences between the two beam models. Vibration suppression strategies from wave point of view are addressed.

10:15

**2aSAb2. A high-frequency theory for thermo-viscoelastic beams.** G. Askar Altay (Dept. of Civil Eng., Bogazici Univ., Bebek 80815, Istanbul, Turkey) and M. Cengiz Dokmeci (Istanbul Univ., Istanbul, Turkey)

To predict the macromechanical response of beams of uniform cross section, a system of 1-D equations is consistently formulated in both differential and variational forms within the frame of the 3-D theory of thermo-viscoelasticity. First, the basic equations of linear, nonisothermal 3-D theory of thermo-viscoelasticity are expressed in terms of the Laplace transformed field variables by a recent variational principle [Altay and Dokmeci, *Acta Mech.* **143**, 91-111 (2000)]. Next, the temperature increment and displacement fields are expressed by the series expansions of the aerial coordinates of beam cross section. Then, a system of higher-order beam equations is derived by means of the variational principle together with the series expansions of field variables. The system of equations governs the extensional, flexural, torsional, as well as coupled vibrations of thermo-viscoelastic beams at both high and low frequencies. Certain cases involving special types of vibrations, geometry, and material properties are recorded. The uniqueness is investigated in solutions of the system of beam equations. [Work supported in part by TUBA.]



**2aSAb3. Special properties of higher-order symmetric Lamb waves on plates.** Michael F. Werby (NRL Code 7180, Stennis Space Center, MS 39529 and Dept. of Phys., Catholic Univ. of America, Washington, DC 20064) and H. Überall (Catholic Univ. of America, Washington, DC 20064)

Exact linear elastodynamic theory is used to derive expressions for all of the modes on a two-dimensional unloaded plate in an effort to understand certain features generated from the symmetric modes on plates and shells. In that context we explain the unusual nature of all  $S_1$  modes including the negative slope of the phase velocity with respect to wave number at critical frequency, the amplitude modulated nature of the transient solution in that frequency zone, and the fact that one type of elastic material has an infinite number of such unusual modes (materials with a Poisson ratio of exactly  $1/3$ ) while the remaining class only appears to have that property for the  $S_1$  mode. We review the asymptotic nature of all of the plate modes and show that the symmetric Lamb modes always have a plateau region in phase velocity relative to wave number and how many of the properties of the symmetric Lamb modes may be used for classification and detection of shells and plates.

**2aSAb4. Reflection of sh waves from an irregular free-end in a semi-infinite elastic plate.** Nahil A. Sobh (Theoret. and Appl. Mech., Univ. of Illinois-Urbana-Champaign, Urbana, IL 61801) and Yagoub N. Al-Nassar (KFUPM, Dhahran 31261, Saudi Arabia)

The problem of anti-plane-wave reflection off an inclined free-end in a semi-infinite elastic plate is investigated. An artificial internal boundary is constructed which divides the plate into two regions. One region contains the irregular free-end while the other contains the rest of the semi-infinite plate. Each admits a separable solution of the wave equation and satisfies all its physical boundary conditions. The full wave solution is finally constructed by matching displacement and stress continuities at the internal boundary. A number of numerical studies illustrate the variation of the reflected modal energy as a function of the free-end inclination and the incident modal energy.

**2aSAb5. An energy finite element formulation for high-frequency vibration analysis of plate structures subjected to heavy fluid loading.** Weiguang Zhang and Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Rd., Ann Arbor, MI 48109-2145)

An energy finite element formulation for computing the high-frequency behavior of plate structures in contact with a dense fluid is presented. The heavy fluid loading effect is incorporated in the derivation of the governing differential equations and in the computation of the power transfer coefficients between plate members. The validity of the new formulation is demonstrated through comparison to an analytical solution and to results computed by conventional finite element models comprised by a large number of elements. The influence of the heavy fluid on the dynamic response of two marine structures is identified by comparing results computed with and without the effect of fluid loading. The impact of the heavy fluid loading on the computation of the power transfer coefficients is also identified. [Research supported by the Michigan Seagrant.]

**2aSAb6. 2.5-D FEM for cochlear-based transducers: Theory and experiment.** John M. Dodson and Karl Grosh (Mech. Eng. Dept., Univ. of Michigan, 3124 G. G. Brown Lab., 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@engin.umich.edu)

The mammalian cochlea is an extraordinary organ exhibiting outstanding sensitivity and frequency discrimination capabilities. This tiny organ senses acoustic input via a coupled pressure-displacement wave along a pair of ducts separated by a partition known as the basilar membrane

(modeled as a variable impedance plate) which exhibits spatial-frequency dependent resonances. The cochlea provides a template for designing devices which utilize this unique mode of transduction and holds potential for a new class of transducer. A reduced-order hybrid analytical/finite-element method (FEM) is used to allow computation of this coupled fluid-orthotropic variable width plate interaction problem. In this method, a 3-D FEM problem is reduced to 2-D discretization by utilization of a limited set of analytical pressure/structural modes orthogonal to the direction of wave propagation. Simple and complex assumed structural mode shapes were investigated to predict the plate response. Comparison to experiments revealed that the deflection of plate supporting structures is important to achieve a good correlation between theory and experiment. The validation program was conducted for isotropic and orthotropic plates mounted on a fluid-filled duct excited by a shaker-driven piston. Design studies for a cochlear-based transducer will be presented.

**2aSAb7. Dynamic response of *in vacuo* prolate spheroidal shells excited by surface force fields.** Sabih I. Hayek (Dept. of Eng. Sci. and Mech., Penn State Univ., University Park, PA 16802-6812, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (Naval Undersea Warfare Ctr. Div. Newport, Newport, RI 02841)

The equations of motion for nonaxisymmetric vibration of prolate spheroidal shells of constant thickness were derived using Hamilton's principle. The thin shell theory used in this derivation includes three displacements and two changes of curvature. The effects of membrane, bending, shear deformations, and rotatory inertias are included in this theory. The resulting five partial differential equations are self-adjoint and positive definite. The shells are excited by axisymmetric line forces. The axisymmetric modal solutions are expanded in an infinite series of comparison functions. These include associated Legendre functions in terms of the prolate spheroidal angular coordinate. Numerical results for the frequency response were obtained for several shell thickness-to-length ratios ranging from 0.005 to 0.1, and for various diameter-to-length ratios, including the limiting case of a spherical shell. Sample mode shapes were obtained at selected resonant frequencies. [Work supported by ONR and the Navy/ASEE Summer Faculty Program.]

**2aSAb8. Recursive Green function technique applied to acoustic wave propagation.** Mauro S. Ferreira, Gerrit E. W. Bauer (Dept. of Appl. Phys., Delft Univ. of Technol., Lorentzweg 1, 2628 CJ Delft, The Netherlands), and Kees Wapenaar (Delft Univ. of Technol., 2628 RX Delft, The Netherlands)

Guided by similarities between electronic and classical waves, a numerical code based on a formalism proven to be very effective in condensed matter physics has been developed, aiming to describe the propagation of elastic waves in stratified media (e.g., seismic signals). This so-called recursive Green functions technique is frequently used to describe electronic conductance in mesoscopic systems. It follows a space discretization of the elastic wave equation in frequency domain, leading to a direct correspondence with electronic waves traveling across atomic lattice sites. An inverse Fourier transform simulates the measured acoustic response in time domain. The method is numerically stable and computationally efficient. Moreover, the main advantage of this technique is the possibility of accounting for lateral inhomogeneities in the acoustic potentials, thereby allowing the treatment of interface roughness between layers. Without those lateral variations, wave propagation across perfectly layered media with planar sources are equivalent to 1-D environments, and therefore should present localization effects. Inclusion of those variations breaks the symmetry and turns the wave propagation into a truly 2-D or 3-D problem, where localization is weaker or may not occur. Numerical techniques without lateral inhomogeneities may inadvertently overestimate localization effects.

## Session 2aSC

## Speech Communication and Psychological and Physiological Acoustics: Special Session Honoring Harry Levitt

Arlene C. Neuman, Chair

*Center for Research in Speech and Hearing, Graduate Center, City University of New York, 365 Fifth Avenue, New York, New York 10016*

Chair's Introduction—8:00

*Invited Papers*

8:05

**2aSC1. Adaptive procedures in psychoacoustics and speech research.** Marjorie Leek (Army Audiol. & Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307-5001, marjorie.leek@na.amedd.army.mil)

Over the past 30 years, adaptive procedures have become ubiquitous in psychoacoustic and speech perception research. The increasing use of these procedures may be traced in part to Harry Levitt's paper describing the flexibility of up-down testing methods to determine the point on a psychometric function associated with a given level of performance [H. Levitt, *J. Acoust. Soc. Am.* **57**, 55–62 (1971)]. Most psychoacousticians have, at one time or another, consulted this article to inform their experimental procedures. However, Harry's contributions to psychometric testing do not begin or end with the publication of this seminal article. Here I will discuss the far-reaching impact of the transformed procedures that formed the basis of the 1971 article, point out some caveats to be considered when using these methods, and review other developmental work that Harry has accomplished to improve psychometric testing, including adaptive methods of fitting hearing aids and of testing speech recognition. [Work supported by NIH (DC00626).]

8:30

**2aSC2. Harry Levitt legacies: Studies of speech perception and intelligibility in children with cochlear implants.** Emily A. Tobey (Callier Adv. Hearing Res. Ctr., 1966 Inwood Rd., Dallas, TX 75235, etobey@utdallas.edu) and Ann Geers (Central Inst. for the Deaf, St. Louis, MO)

Harry Levitt has contributed a substantial body of knowledge regarding the speech perception and production skills of hearing-impaired children. In this paper, Dr. Levitt will be honored by describing the speech perception, speech intelligibility, and acoustic characteristics of 136 cochlear implanted children between the ages of 8 and 9 who have at least 4 years experience with a Nucleus multichannel cochlear implant. Half of the children used auditory-oral modes of communication and the other half used total communication. The questions addressed are as follows. (a) Do children implanted before 5 years of age develop intelligible speech? (b) Are speech perception skills reflected in the acoustic measurements of speech in children with cochlear implants? (c) What factors are associated with intelligible speech in children with cochlear implants? The presentation will describe the population demographics, speech perception measures (including open and closed set speech perception performance), and measures of speech production (including speech intelligibility, communication breakdowns, parental rating scales, and acoustic characteristics). Factors contributing to high levels of speech perception and intelligibility will be described and contrasted. Implications for the development of intelligible speech will be discussed, particularly as it relates to mode of communication. [Work supported by NIH-NIDCD.]

8:55

**2aSC3. Reordering disordered speech: The search for speech prostheses.** H. Timothy Bunnell (duPont Hospital for Children and Univ. of Delaware, 1600 Rockland Rd., Wilmington, DE 19803)

The development over the last quarter century of computer-based speech processing has allowed investigators to (a) modify disordered speech as a means of gaining a better understanding of its nature, and (b) look toward the possibility of altering disordered speech to enhance its perceived naturalness or its intelligibility. Harry Levitt and his colleagues [H. Levitt, *IEEE Trans. Audio. Electroacoust.* **21**, 269–273 (1973); M. J. Osberger and H. Levitt, *J. Acoust. Soc. Am.* **66**, 1316–1324 (1979)] were among the first to examine the possibility of enhancing disordered speech via computer speech processing. Since then, a number of other investigators have taken up this line of research, employing a variety of speech processing techniques to examine issues raised by Levitt (1973), particularly regarding the relative importance of temporal versus spectral distortions in determining the intelligibility or naturalness of disordered speech. In this work, I will trace some of the history of this line of research, including work that Grace Yeni-Komshian and the author have conducted [*J. Acoust. Soc. Am.* **104**, 637–647 (1998)], and subsequent related studies.

**2aSC4. Harry Levitt and hearing aids.** Jont B. Allen (AT&T Labs, Rm. E161, Florham Park, NJ 07974)

Levitt has always been interested in technology transfer. In 1980 he became a proponent of digital hearing aids and fitting systems. I first got to know Levitt during the 1983 AT&T venture into a multiband compression hearing aid, proposed to Waldhauer by Villchur, based in the ideas of Steinberg and Gardner (1938). One of the first things we did was to visit Levitt and pepper him with questions. He saw it as a huge opportunity, but it was difficult because he was consulting for Nicolet. Eventually AT&T decided to return to its "core" business, and the venture was sold to Resound Corp. At CUNY we also developed the loudness growth in 1/2 octave bands (LGOB) test, which was popular at one time. Soon after, Levitt convinced me to port our research distortion product analyzer, called CUB<sup>o</sup>DIS, to the PC platform. It was then further developed by Jeng (Levitt's student), and is presently Starkey's DPOAE/TOAE DP-2000 infant hearing screener. All this technology transfer would not have taken place if it were not for Harry Levitt—so for that we must graciously thank him.

**2aSC5. Acoustic and psychoacoustic benefits of adaptive compression thresholds in hearing aid amplifiers that mimic cochlear function.** Julius L. Goldstein (Washington Univ., Campus Box 1127, St. Louis, MO 63130), Michael Valente (Washington Univ. Med. School, St. Louis, MO 63110), and Roger D. Chamberlain (BECS Technol., Inc., St. Louis, MO 63132)

Automatic gain control of linear amplifiers dominates advanced hearing aid design, and has been extensively studied [Levitt *et al.*, IEEE (1980); Dillon, *Ear Hear.* **17**, 287–307 (1996)]. The normal cochlea uses essentially nonlinear, rapidly compressive amplification under efferent control [Kiang *et al.*, *Hear. Res.* **22**, 171–182 (1986); Robles *et al.*, *J. Acoust. Soc. Am.* **80**, 1364–1374 (1986)], whose salient characteristics have been modeled [Goldstein, *Hear. Res.* **49**, 39–60 (1990)] and are presently adopted for multichannel hearing aids. Signal transduction in each channel is linear at low and high sound pressure levels (SPL), and smoothly joined by a compressive range. The transition from linear to compressive response at low SPL is controlled by an adaptive compression threshold, whose quiescent level provides the gain needed for weak sounds. For sustained sounds at higher SPL the compression threshold adapts upwards, providing effectively linear response and better acoustic quality for speech in noise. Pilot psychoacoustic experiments with a design simulation demonstrated that both normal and impaired hearing subjects perceive the improved quality, and comprehend speech in noise at least as well as with advanced hearing aids [Valente *et al.*, *J. Am. Acad. Audiol.* **9**, 342–360 (1998)]. [Work supported by NIDCD.]

**10:10–10:25 Break****2aSC6. Feedback cancellation in hearing aids.** James M. Kates (AudioLogic/Cirrus Logic, 2465 Central Ave., Ste. 100, Boulder, CO 80301, jkates@audiologic.cirrus.com)

Harry Levitt was an early advocate of digital signal processing for hearing aids. His interest included not only algorithms for compression and speech enhancement, but also more practical issues such as obtaining the desired hearing aid versus frequency response at the ear drum. Acoustic feedback is one problem that can limit the maximum gain possible in a hearing aid. Feedback cancellation, in which the acoustic feedback signal is estimated and subtracted from the microphone input, allows for greater hearing-aid signal amplification, and feedback cancellation was included in the work that Harry supported in his research group. In this presentation, the effects of feedback on the hearing-aid response will be reviewed. Digital adaptive filter techniques for feedback cancellation will then be presented, along with measurements indicating the limitations of feedback cancellation.

**2aSC7. Speech intelligibility and listener preferences for amplitude-compressed speech: Speech-based STI predictions and measurements on listeners with simulated hearing loss.** Peninah F. Rosengard, Louis D. Braida (Res. Lab of Electron., MIT, Cambridge, MA 02139), and Karen L. Payton (Univ. of Massachusetts–Dartmouth, North Dartmouth, MA 02747)

Relations between objective intelligibility scores, subjective pleasantness ratings, and estimates of the STI for speech processed by multi-band amplitude compression systems were studied in normal-hearing listeners with simulated hearing loss. STI estimates were based on modulation spectrum changes in the processed speech signals [Payton and Braida, *J. Acoust. Soc. Am.* **106**, 3637–3648 (1999)]. Linear amplification and two syllabic compression conditions were tested with and without two backgrounds: Speech-spectrum noise and restaurant babble. Signals were compressed independently in four nonoverlapping frequency bands with compression ratios of two and three, and attack and release times of 20 and 200 ms, respectively. The NAL-R formula determined output frequency-gain characteristics. Flat, 50 dB, sensorineural hearing losses were simulated in normal-hearing listeners via multiband expansion [Duchnowski and Zurek, *J. Acoust. Soc. Am.* **98**, 3170–3181 (1995)]. Speech intelligibility and pleasantness ratings were obtained. All conditions were also evaluated using the modified STI. The STI predicted speech in restaurant babble would be more intelligible than speech in noise for each compression condition. The data reflected this for the two compression conditions. The STI predictions were also consistent with pleasantness ratings: Linear amplification was the most pleasant compression condition. [This work supported by NIDCD.]



**2aSC8. Recent advances in audio and video text-to-speech synthesis.** Ann K. Syrdal (AT&T Labs Res., 180 Park Ave., Florham Park, NJ 07932), Joern Ostermann, Eric Cosatto, and Hans Peter Graf (AT&T Labs Res., Middletown, NJ 07748)

Harry Levitt characteristically combines a statistical or data-driven engineering approach with extensive scientific knowledge when tackling fundamental research questions (such as how to measure co-articulation) and practical problems (such as evaluating intelligibility). Some of Harry's expansive research interests and contributions have focused on audio and video text-to-speech (TTS) synthesis. Some major recent improvements in these technologies that have resulted from applying similar approaches are discussed. The unit selection method of synthesis described differs from both the earlier techniques of formant synthesis and of concatenative synthesis from a diphone inventory. In unit selection synthesis, speech or video units are selected from among multiple candidates in a large database by optimizing a cost function. The cost functions used for unit selection are estimates of perceptual distances. The improvement in naturalness for both audio and visual TTS achieved through the unit selection technique is demonstrated, and some of its current limitations and future work needed in this area is discussed.

### *Contributed Paper*

11:40

**2aSC9. Adaptation algorithms for noise reduction in hearing aids based on auditory models.** Birger Kollmeier, Jürgen Tchorz, Thomas Wittkop, and Volker Hohmann (Medizinische Physik, Universität Oldenburg, D-26111 Oldenburg, Germany, birger.kollmeier@uni-oldenburg.de)

Noise reduction algorithms for hearing aids usually employ certain assumptions about the ambient noise and the target (speech) signal that are only met in certain acoustical situations. In order to reduce unwanted artifacts whenever the acoustical real-life situation does not meet these assumptions, two approaches are introduced and tested that classify the acoustic environment. Both approaches are motivated by auditory models

and are used to control the parameters of noise-reduction schemes. The first approach uses the amplitude modulation spectrogram (AMS) in combination with a neural net to estimate the speech-to-noise ratio either on the broadband signal or within narrow bands. It can be used for single-channel (i.e., monaural) noise reduction even for fluctuating background noise and unfavorable signal-to-noise ratios. The second approach derives the "degree of interaural coherence" from the input signals to both ears that decreases with increasing reverberation and increasing number of active sound sources. It is combined with several binaural noise reduction techniques, i.e., directional filtering, dereverberation, and adaptive beamforming. The advantage of using these approaches will be presented and discussed on the basis of speech intelligibility and subjective quality assessment data.

TUESDAY AFTERNOON, 5 JUNE 2001

RED LACQUER ROOM, 1:00 TO 5:00 P.M.

### **Session 2pAA**

#### **Architectural Acoustics: Halls for Music Performance . . . Another Two Decades of Experience 1982–2002**

Christopher A. Storch, Cochair

*Artec Consultants, Inc., 114 West 26th Street, 9th Floor, New York, New York 10001*

Timothy J. Foulkes, Cochair

*Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776*

Ian B. Hoffman, Cochair

*The Talaske Group, Inc., 105 North Oak Park Avenue, Oak Park, Illinois 60301*

Posters from Session 2aAAa will remain on display until 5 p.m.

## Session 2pAB

## Animal Bioacoustics and Psychological and Physiological Acoustics: Comparative Aspects of Auditory System Development

Andrea M. Simmons, Chair

Department of Psychology and Neuroscience, Brown University, Box 1853, Providence, Rhode Island 02912

Chair's Introduction—1:15

## Invited Papers

1:20

**2pAB1. Confocal imaging of sensory organ formation during *Xenopus* inner ear development.** Elba E. Serrano and Quincy A. Quick (Biol. Dept., New Mexico State Univ., Las Cruces, NM 88003)

During *Xenopus* larval development, the otic vesicle gradually gives rise to several auditory and vestibular sensory organs that are innervated by the eighth cranial nerve. Confocal imaging methods were used to examine the appearance of stereociliary bundles in the various sensory epithelia during *Xenopus* development (stages 28–50), and to gather comparative developmental data that show how the inner ear gradually forms distinct auditory and vestibular compartments discernable by larval stage 50. A BioRad MRC-1024 confocal microscope was used to collect digitized images from inner ears labeled with Alexa 488 phalloidin to detect the actin cytoskeleton, and propidium iodide to highlight DNA in cell nuclei. Confocal images from whole inner ears clearly show the developmental emergence of uniquely organized sensory epithelia populated by hair cell bundles of varying morphology, as well as the position of the sensory ganglia relative to the endorgans. Images from sectioned tissue provide additional information about cell morphology and the arrangement of the sensory epithelium in relation to neural innervation. Taken together, the image data illustrate the complex structural changes that underlie inner ear organogenesis and morphogenesis. [Work supported by grants to EES (NIH NIGMS, NASA) and awards to QQ (NASA NMSGC Fellowship, NIGMS RISE).]

1:40

**2pAB2. Developmental changes in GABA expression across metamorphosis.** Andrea M. Simmons, Seth S. Horowitz, and Judith A. Chapman (Dept. of Psych. and Neurosci., Brown Univ., Providence, RI 02912, andrea\_simmons@brown.edu)

Gamma-aminobutyric acid (GABA) expression in the developing brain has been implicated in factors such as process outgrowth, cell proliferation, cell migration, and synaptogenesis. Immunofluorescent staining techniques were used to characterize changes in GABA distribution in medullary and cerebellar regions of the bullfrog's brainstem across metamorphosis. Auditory and vestibular nuclei show profound developmental changes in distribution; the cerebellum and cerebellar nuclei show little change, being strongly stained at all stages examined. In early stage tadpoles, the dorsolateral nucleus (DLN), vestibular nucleus (VN), and anterior lateral line (LLa) nucleus show widespread punctate and diffuse label, but few discrete GABAergic cells. Little or no staining is observed in the superior olivary nucleus (SON). This basic pattern persists until just before metamorphic climax stages, when discrete GABAergic cells first begin to appear in the SON; more discrete cell staining is also visible in the DLN and VN. At metamorphic climax, the DLN, VN, and SON show clear populations of GABAergic cells. In postmetamorphic froglets and adults, GABAergic cells and GABA-labeled fibers exist in sharply delineated regions of these nuclei, with limited diffuse/puncta label. These changes are consistent with the idea that GABA plays important roles in regions of the brainstem undergoing developmental transformation.

2:00

**2pAB3. Experience and auditory brainstem development.** Edwin W. Rubel (VMB Hearing Res. Ctr., Box 357923, Univ. of Washington, Seattle, WA 98195, rubel@u.washington.edu)

Understanding the cellular basis of experiential influences on auditory system development involves characterizing (1) the inter-cellular signaling molecule(s) responsible for activity-based long-term changes in targeted neurons; (2) the cascade of cellular events which alters the phenotype of the targeted neurons; and (3) the biological basis of developmental periods of enhanced brain plasticity, critical periods. I will summarize a series of experiments examining neuronal integrity in the cochlear nucleus of birds and mammals following manipulations of eighth-nerve activity. Comparisons of conductive hearing loss and TTX blockade of eighth-nerve activity along with *in vitro* experiments indicate that glutamate release and activation of group 1 metabotropic glutamate receptors (mGluRs) are essential for normal survival of cochlear nucleus neurons. The cascade of intracellular events determining survival or death of

2p TUE. PM

cochlear nucleus neurons involves regulation of intracellular free calcium by one or more mGluRs. Finally, new studies using transgenic mice are beginning to reveal biological mechanisms responsible for developmental changes in cellular responses to afferent deprivation. I will present recent data showing that manipulations of bcl-2 gene expression dramatically alter the critical period of susceptibility to afferent deprivation. [Research supported by grants from NIH/NIDCD.]

2:20

**2pAB4. Thyroid hormone influences on the development of auditory function in mammals.** Edward J. Walsh and JoAnn McGee (Boys Town Natl. Res. Hospital, 555 N. 30 St., Omaha, NE 68131)

The essential role of the thyroid hormone in development is one of the most clearly established principles of biology. Although its importance as a factor affecting the maturation of the inner ear is unquestionable, the precise mode of its actions and the identity of its intracochlear targets remain issues of conjecture. Although both alpha1 and beta forms of the thyroid hormone receptor (TR) are expressed in the developing cochlea of mammals, permanent auditory deficits occur only when TR-beta1, which is distributed widely in the brain and inner ear, is defective. The presence of multiple forms of thyroid hormone receptors with different functions suggests that the mode of thyroxins action in the inner ear is complex. When thyroxine is unavailable during development, as in Tshr (hyt) mutant mice or animals rendered hypothyroid by treatment with goitrogens, a distinctive constellation of functional deficits are observed, suggesting that the primary, enduring locus of pathology lies within the domain of the cochlear amplifier. We will discuss the physiological basis underlying this position and propose a mechanism that accounts for the phenomenon, as well as the wide range of anatomical abnormalities that have been observed in the cochleae of hypothyroid mammals. [Work supported by NIDCD grant DC00982.]

2:40

**2pAB5. Development of perceptual dimensions lags the ability to perceive details.** Lincoln Gray, Joshua Breier (Univ. of Texas, Houston Med. School, 6431 Fannin St., Houston, TX 77030), Randy Diehl (Univ. of Texas, Houston, TX 77030), Peggy Haardt, and Christine Turley (Univ. of Texas Medical Branch, Houston, TX 77030)

Studies with humans and animals show that the ability to discriminate fine changes along an auditory continuum develops before the ability to appreciate the continuum. The visual analogy would be seeing details before the "big picture." This trend is discussed with data from normal newborn chickens responding to changes in frequency and azimuth, from patients after corrected congenital hearing loss, and from children and chickens exposed to lead. In each instance, subjects respond to small changes before they respond consistently to large changes. Multidimensional scaling can reveal the "attribute" or perceived dimension that distinguishes a set of stimuli. The attribute is compared to the difference limen. These data suggest that discriminations can emerge before the appropriate perception of a sensory attribute. Perceptual development may not always involve a process of increasingly fine discriminations along continua that are established at an early age, but instead may involve the ability to process global pattern. Some theories suggest that perceptions "sharpen" over development. Sharpening does not seem to occur in situations we have studied: over normal development, after early deprivation, or after early lead poisoning. If confirmed in other situations, this could suggest novel approaches to ameliorate some early perceptual processing deficits.

3:00

**2pAB6. Neural representation and perception of speech sounds in children with learning problems.** Nina Kraus (Depts. Commun. Sci., Neurobiology, Otolaryngol., Northwestern Univ., 2299 N. Campus Dr., Evanston, IL 60201) and Cynthia King (Northwestern Univ.)

An inability to process auditory information, especially speech, characterizes many children with learning and attention problems. These speech-sound perception problems may arise, at least in some cases, from faulty representation of the speech signal in central auditory centers. Our working hypothesis is that acoustic-phonetic disorders are abnormalities in preconscious neurophysiologic representation of sound structure by central auditory pathway neurons and is reflected by subcortical and cortical aggregate neural responses. Brainstem and cortical potentials (ABR, FFR, cortical P1/N1/N2, MMN) reflect activity from different anatomic sources and represent different aspects of auditory function. An obvious characteristic distinguishing these responses is their development time course. Results indicate that while normal and learning impaired children do not differ in auditory system development per se, they develop different listening strategies which affect the neural representation of sound structure along the auditory pathway. [Work supported by NIH RO1 DC 01510.]

### *Contributed Papers*

3:20

**2pAB7. The auditory system as a window onto the processes of prenatal development.** Dennis McFadden (Dept. of Psych. and Inst. for Neurosci., Mezes Hall 330, Univ. of Texas, Austin, TX 78712)

Otoacoustic emissions (OAEs), auditory evoked potentials (AEPs), and hearing sensitivity all exhibit sex differences of several types in humans, and these sex differences are highly similar in newborns and adults. The strong implication is that these differences originate during prenatal development and persist through life. The most parsimonious explanation

is that these sex differences are produced by androgenic processes operating on the ear and brain of the developing fetus. Certain special populations of humans have been shown to have OAEs and/or AEPs that differ from the sex-typical values, and again the most parsimonious explanation is that the degree of androgen exposure is the basis for these differences. Certain nonauditory measures from these special populations are in accord with this interpretation. The data from special populations will be summarized and a number of alternative explanations considered. Implications for research on the development of hearing will be discussed. [Work supported by NIDCD.]

**2pAB8. Pattern adaptation in midbrain and medullar auditory units of the frog.** Nikolay G. Bibikov (N. N. Andreyev Acoust. Inst., Shvernik St. 4, Moscow 117036, Russia)

The responses of medullar and midbrain auditory cells to prolonged amplitude-modulated tone stimulation were recorded extracellularly in the dorsal medullar nucleus, superior olive and torus semicircularis of curarized grass frogs (*Rana temporaria*). The majority of the cells with tonic response to tone bursts showed a significant adaptation in their firing rate within first 20 to 30 s and then stabilized gradually. The temporal course of the rate adaptation was approximated by a single or double exponent plus a steady component corresponded to sustained firing. The dependence of the time constant of the rate decrease and the sustained firing value upon mean carrier intensity, modulation depth and modulation frequency have been studied for 128 medullar and 105 midbrain units. Generally, adaptation was stronger for midbrain versus medullar units. In each auditory nuclei the adaptation decreased with the increase in carrier intensity and modulation depth. The weakest adaptation (the highest value

of sustained rate) was usually observed when a low-frequency noise was used as a modulating waveform. The comparison to the mammalian central auditory units is discussed.

3:50

**2pAB9. The fish swimbladder as an acoustic waveguide.** David T. I. Francis (School of Mathematics and Statistics, Univ. of Birmingham, Edgbaston, Birmingham B15 2TT, UK) and Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

The possible role of the fish swimbladder as a sound-sensing organ is investigated using numerical models based on the boundary-element method. A number of gadoid specimens are considered. For each specimen, wire-frame models of the swimbladder surface are derived from morphometric data. The response of each swimbladder to a continuous incident wave is calculated at discrete frequencies up to 50 kHz. Pressure distributions within the swimbladders and velocity distributions on the surfaces are presented. Results reveal high-pressure and high-velocity regions around the anterior horns of the swimbladders. [This work originated in European Community RTD—Contract No. MAS3-CT95-0031 (BASS).]

TUESDAY AFTERNOON, 5 JUNE 2001

SALONS III/IV, 1:00 TO 5:30 P.M.

### Session 2pBB

## Biomedical Ultrasound/Bioresponse to Vibration and Signal Processing in Acoustics: Novel Imaging Techniques in Biomedical Ultrasound

Christy K. Holland, Chair

*Department of Radiology, University of Cincinnati, 234 Goodman Street, Cincinnati, Ohio 45219-2316*

Chair's Introduction—1:00

### Invited Papers

1:05

**2pBB1. 3-D compound imaging with refraction and motion correction.** Paul L. Carson, Jochen F. Kruecker, Gerald L. LeCarpentier, Charles R. Meyer (Dept. of Radiol., Univ. of Michigan, Kresge III, R3315, Ann Arbor, MI 48109-0553), and J. Brian Fowlkes (Univ. of Michigan, Ann Arbor, MI 48109-0553)

The early goal of multiview imaging was delineation of quasi-specular image boundaries. Medical compound imaging was partially replaced because of resolution loss by refraction, misregistration, and tissue motion, and completely replaced due to incompatibility with real-time imaging. Three potential improvements in: boundary delineation, volumetric contrast-to-noise ratio, and Doppler anisotropy reduction (BC&D) have remained an attraction of compounding. Now marketed, in-plane compounding in real-time and in extended field of view imaging offer the potential for in-plane, image-based reregistration of multiple views. Image volume-based registration (IVBaR) of separate angular views in 3-D data sets allows full 3-D displacement corrections. Compounding out of the image plane (elevational compounding) increases the number of independent views, increasing potential yield from improvements BC&D. Our work has been in elevational compounding with conventional linear arrays, in contact with, and partially moving the proximal tissues. IVBaR using the mutual information metric and thin plate spline interpolation of warped volumes has produced exceptional reclamation of spatial resolution along with the expected improvements BC&D. Full-volume, warped registration, and a faster, subvolume registration have been compared *in vivo* and with realistic beam aberrations in test objects. [Work supported in part by PHS Grant No. R01HL54201 from the NHLBI.]

**2pBB2. Artifact reduction in medical ultrasound.** Gary A. Schwartz (ATL Ultrasound, P.O. Box 3003, Bothell, WA 98042, gary.schwartz@philips.com)

Interpretation of medical ultrasound images is confounded by the presence of significant acoustic imaging artifacts. These artifacts result from diffraction, propagation, and scattering effects and regional variations in these effects. Commercial diagnostic imaging systems have taken various design approaches to mitigate the artifacts. Anatomic features widely vary in size relative to the imaging wavelengths, resulting in angular scattering variations and speckle-limited resolution. Tissue attenuation varies greatly resulting in shadowing and enhancement. This review will look at the origins of certain acoustic imaging artifacts and discuss methods that have been applied to address their contribution to image quality. Included in the discussion will be acoustic methods (such as spatial compounding), and signal processing methods (such as attenuation compensation and frequency compounding). Design approaches and clinical image examples will be presented.

**2pBB3. Higher order nonlinear ultrasonic imaging.** Bruno Haider (GE Corporate R&D, KW C1315, One Research Circle, Niskayuna, NY 12309) and Richard Y. Chiao (GE Medical Systems, Milwaukee, WI 53219)

The processing of second harmonic echoes from both biological tissue and contrast agents has generated new diagnostic methods in medical ultrasound. The work presented here demonstrates the extraction of higher order nonlinearities. The underlying idea is to model the nonlinear wave propagation or reflection from a contrast bubble by a polynomial expansion of some basis waveform. When this model is excited by a number of transmit pulses which only differ in their amplitude and phase then the coefficients of this polynomial model can be extracted through least squares inversion. The coefficients correspond to the individual nonlinear components. An important feature of the method is the evaluation of nonlinear components whose spectra are folded back into the transmission band. All odd order nonlinearities can create such echo components. The reception of these components eliminates the high bandwidth requirements encountered in second harmonic imaging. Higher-order even harmonics may also be detected by taking advantage of the harmonic fold-back process. Folded frequency components will be centered around DC and at two times the transmit frequency ( $2f_0$ ). This still requires a bandwidth sufficient to detect signals at  $2f_0$  but eliminates the reception at higher multiples of  $f_0$ .

**2pBB4. B-mode blood flow (B-Flow) imaging.** Richard Y. Chiao (GE Medical Systems, P.O. Box 414, Milwaukee, WI 53201)

B-Flow is a new technique that extends the resolution, frame rate, and dynamic range of B-mode to simultaneously image blood flow and tissue. B-Flow relies on coded excitation to boost weak signals from blood scatterers and on tissue equalization to simultaneously display flowing blood and tissue without threshold decision and overlay. Various classes of codes such as Barker and Golay may be used. Clinical B-Flow cine-loops demonstrate  $3\times$  resolution and frame rate improvement over color flow, which, together with over 60 dB of display dynamic range, is able to image hemodynamics and vessel walls with unprecedented clarity.

### *Contributed Papers*

**2pBB5. A two-dimensional amplitude-steered array for real-time volumetric imaging.** Catherine H. Frazier (Johns Hopkins Appl. Phys. Lab., 11100 Johns Hopkins Rd., Laurel, MD 20723), W. Jack Hughes (Penn State Univ., State College, PA 16801), and William D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL 61801)

Real-time three-dimensional acoustic imaging is difficult in water or tissue because of the medium's slow sound speed. Conventional pulse-echo data acquisition, which uses at least one transmit pulse per line in the image, does not allow for the real-time update of a volume of data at practical ranges. Recently, we presented a linear amplitude-steered array [J. Acoust. Soc. Am. **107**, 2430 (2000)], based on an earlier concept [J. Acoust. Soc. Am. **59**, 1040 (1976)], that allows the collection of a plane of data with a single transmit pulse by spatially separating frequencies in the lateral direction. The linear-phased amplitude-steered array uses frequency separation to determine vertical position and conventional beamforming to determine horizontal location. To image a volume of interest with a single transmit pulse, the received signal must contain information to give vertical, horizontal, and range position of the target. The target range is obtained from the time elapsed until the reflected signal is received. The

vertical position information is determined by the returned signal's frequency. The target's horizontal position is found by using conventional, linear phased array processing. In this study, we describe the volumetric imaging system, giving the two-dimensional array design, and describing data acquisition and image processing strategies. [Work supported by DARPA's Sonoelectronics Program.]

**2pBB6. Ultrasound image based on ultrasound characterization of tissue microstructure of spontaneous rat mammary tumors.** Michael L. Oelze, William D. O'Brien, Jr. (Dept. of Elec. and Computer Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, oelze@brl.uiuc.edu), and James F. Zachary (Univ. of Illinois, Urbana, IL 61802)

Experimental results of acoustic backscatter from spontaneous mammary tumors in rats are obtained over the frequency range of 1 to 15 MHz. The power spectrums (dB) from the experimental results are compared to theory in order to obtain information about tissue microstructure. The theoretical power spectrum assumes an isotropic 3-D Gaussian function as the spatial autocorrelation function used to describe the distribution of scatterers in the tissues. The theoretical power spectrum can be shown for



small scatterers (diameter less than 0.2 mm) to be proportional to the Rayleigh spectrum times a second-degree polynomial in frequency. Independent values of the average scatterer diameter and scatterer concentration are obtained from the coefficients of the second-degree polynomial. Scatterer sizes and concentrations measured by ultrasound backscatter are compared to histological data. B-mode images are made of the tumors and surrounding tissues with superimposed regions-of-interest (ROIs) parameterized by the scatterer sizes and concentrations. [Work supported by NIH CA 079179.]

2:55

**2pBB7. Spatial sampling resolution study on BAI-mode imaging for defect detection.** Xiangtao Yin (Dept. of Elec. and Computer Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, xyin@uiuc.edu), Osama Nayfeh, Scott A. Morris, and William D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL 61801)

The principal concern with regard to defect detection in hermetically sealed flexible food packages is the safety and shelf life of the food. The spatial sampling issue of the pulse echo backscattered amplitude integral (BAI) mode imaging technique for defect detection is investigated. In our previous spatial sampling study, it has been shown that for channel defects, the contrast to noise ratio (CNR) degrades as a function of scanning step size on each dimension of the image. To further understand spatial sampling resolution of this method, BAI imaging technique is applied to rectilinear grided dot samples with different grid sizes (distance between adjacent dots). Data is collected with the transducer scanned in a zigzag raster pattern. Quantitatively, the CNR and  $\Delta$ BAI values are assessed to evaluate the image quality versus the changing spatial interval between dots. At a given operating frequency and a fixed spatial grid size, the CNR and  $\Delta$ BAI value degrade as a function of the scanning step size on each dimension. Not only the scanning step size, but also the ultrasound beam spot size affect the spatial sampling resolution in the BAI imaging technique for defect detection. [Work supported by the C-FAR program, University of Illinois.]

3:10–3:30 Break

3:30

**2pBB8. Spatial resolution performance of a one-dimensional linear amplitude-steered array in an attenuating medium.** Sarah A. Hartleben, Virginia Lassale, and William D. O'Brien, Jr. (Dept. of Elec. and Computer Eng., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, hartlebe@uiuc.edu)

The amplitude-steered array concept [J. Acoust. Soc. Am. **59**, 1040 (1976)] introduced the idea of steering the maximum response of a linear array by amplitude weighting the output signals of the elements. Recently, an evaluation of the linear amplitude-steered array assessed its performance such as lateral and axial resolution in a lossless medium [Frazier, Hughes, and O'Brien, J. Acoust. Soc. Am. **107**, 2430 (2000)]. In the present work, the array's performance is evaluated in an attenuating medium similar to biological tissue (1 dB/cm MHz). Similar to the array's performance in a lossless medium, the length of the array limits the axial and lateral resolution in the attenuating medium. As the steering direction is decreased, which corresponds to increasing frequency, the array's performance with attenuation has a worse resolution when compared with its performance without attenuation. For example, using a 10-cm-length array and a 6-deg steering direction (4.6 MHz), the axial resolution ( $-3$ -dB point spread function width) with attenuation was 7.9 mm, compared with 6.4 mm without attenuation. When the steering direction was increased to 12 deg (2.3 MHz), the axial resolution with attenuation was 12.8 mm compared with 11.5 mm without attenuation. In the attenuating medium there is increased importance on the steering direction or frequency. [Work supported by NIH CA09067.]

3:45

**2pBB9. Elasticity imaging with time-resolved pulsed elastography.** Laurent Sandrin, Micka Tanter, Stefan Catheline, and Mathias Fink (Laboratoire Ondes et Acoustique, Supérieure de Physique et Chimie Industrielles de la Ville de Paris, Université Denis Diderot, CNRS UMR 7587, 10 rue Vauquelin, 75005 Paris, France)

Time-resolved pulsed elastography is a promising technique for imaging the shear modulus of soft tissues. It is based on the investigation of a low-frequency transient shear wave using an ultrafast ultrasonic imaging system (up to 10 000 frames/s) composed of 128 channels sampled at 50 MHz and having 2 Mbytes. The system is connected to a linear array of transducers. Displacements induced by the propagating shear wave are measured using cross correlation of the ultrasonic signals. A low-frequency vibrating device was designed. The linear array of transducers is placed between two rods fixed to electromagnetic vibrators. The rods are either orthogonal or parallel to the active surface of the transducer array. The low-frequency shear waves are sent using the rods which are placed in such a way that the lobes of the induced shear waves superimpose in front of the transducer array. Large displacements are observed in the tissues which makes deeper investigations possible. Movies of the low-frequency (50–200 Hz) shear wave propagation through soft tissues can be used either to estimate the shear modulus distribution in the medium by direct local inversion, or to localize visually lesions of unexpected elasticity. In vivo measurements in human breast will be presented and discussed.

4:00

**2pBB10. Quantitative elasticity imaging with ultrasound.** Paul E. Barbone (Dept. of Aersp. and Mech. Eng., Boston Univ., Boston, MA 02215 and Inst. of Cancer Res., Sutton, Surrey, UK) and Jeffrey C. Bamber (Inst. of Cancer Res. and Royal Marsden Hospital, Sutton, Surrey, UK)

For the past 10 or so years, various researchers have proposed different ways to measure strain distributions *in vivo*. The measured strains result from various sources including external palpation, low frequency vibration, or internal motion. By examining the relative strains in adjacent tissues, it is thought that one can infer the relative distributions of tissue stiffness. In this presentation, we discuss the process of creating a quantitative stiffness image from a given measurement of tissue strain. We show that the strain image by itself is insufficient information to infer the elastic stiffness. By examining the well-posedness of the inverse problem, we determine what information is needed to supplement the strain image in order to quantitatively infer the elastic stiffness distribution. Methods for obtaining the required information *in vivo* are currently being developed. We show examples of incorrect and misleading stiffness reconstructions in the absence of the required data. [Work supported by BU, NIH, Fulbright Foundation, ICR, and NSF via CenSSIS ERC.]

4:15

**2pBB11. A new technique for real-time freehand ultrasonic elasticity imaging.** Yanning Zhu, Timothy Hall, and Larry Cook (Dept. of Radiol., Univ. of Kansas Med. Ctr., 3901 Rainbow Blvd., Kansas City, KS 66160, yzhu@kumc.edu)

A method for high-speed, freehand ultrasonic elasticity imaging will be described. The method is based on a modified block matching technique. The modification serves two purposes. First, in order to achieve real-time performance a forward prediction mechanism is used to reduce the computational load that a conventional block matching technique requires. Second, in order to accommodate the waveform decorrelation associated with freehand scanning a statistical error detection and correction method was developed. This prevents the displacement estimation errors from propagating to affect a larger area. This new algorithm has been implemented on a Siemens Elegra (a high-end clinical ultrasound imaging system). The initial tests show that the new system can perform B-mode and strain display task at real-time frame rates. The strain images have low

noise and excellent contrast. Some results obtained with this system will also be presented. [Support by USAMRAA DAMD17-00-1-0596, NSF BES-9708221, and Siemens Medical Systems is gratefully acknowledged.]

4:30

**2pBB12. Comparison of stiffness estimated with vibration elastography to dynamic mechanical analyzer results.** Travis Oliphant (Elec. and Computer Eng. Dept., Brigham Young Univ., Provo, UT 84602), Randy Kinnick, James Greenleaf, Alex Dresner, and Richard Ehman (Mayo Foundation, Rochester, MN 55905)

Several pathological conditions induce altered tissue mechanical properties. The goal of elastography is to noninvasively generate images and/or measurements of tissue mechanical properties. Vibration elastography is a technique for measuring stiffness using low-frequency (10–1000 Hz) shear waves to vibrate a region of interest. Ultrasonic time-delay estimation techniques or phase-contrast magnetic resonance methods measure the amplitude and phase of one or more components of the harmonic vibration everywhere in the region. Under a linear, isotropic, and incompressible tissue model, the displacement field in a locally homogeneous region satisfies the vector-Helmholtz equation and stiffness estimation becomes an inverse-scattering problem where the field is known *inside* the region of interest. Using an optimal inversion algorithm recently developed by the authors and a data-dependent noise-model, an estimate of local stiffness along with its uncertainty is reported at frequencies from 200–600 Hz for three gels of varying concentrations. For each gel, the displacement is measured using either ultrasound or magnetic resonance. The vibration elastography estimate of stiffness for the three gels is less than the complex modulus measured over a frequency range of 10–250 Hz using the shear sandwich clamp on a dynamic mechanical analyzer (DMA 2980, TA Instruments) by an approximate factor of 3.

4:45

**2pBB13. Response on the surface of a viscoelastic medium due to subsurface acoustic sources with application to vascular medical diagnosis.** Yigit Yazicioglu, Thomas J. Royston, and Francis Loth (Univ. of Illinois at Chicago, Chicago, IL 60607, troyston@uic.edu)

The vibratory response at the surface of a viscoelastic material due to buried spherical and cylindrical acoustic sources is considered theoretically, numerically, and experimentally. An analytical solution that accounts for the compression, shear, and surface wave response to the buried source in a halfspace is formulated and compared with numerical finite-element simulations and experimental studies on finite-dimension phantom models. The focus here is on developing a better understanding of how biological soft tissue affects the transmission of vibro-acoustic energy from biological acoustic sources below the skin surface, such as a turbulent regime in an artery caused by a partial blockage. Such an understanding could catalyze the development of noninvasive procedures using an array of inexpensive acoustic sensors on the skin surface for the identification and monitoring of vascular blockages, a precursor to many serious cardiovascular diseases. [Research supported by NIH NCRR Grant No. 14250.]

5:00

**2pBB14. Efficient three-dimensional cylindrical-geometry ultrasound imaging.** Mark A. Haun, Douglas L. Jones (Dept. of Elec. and Computer Eng., Univ. of Illinois, 1308 W. Main, Urbana, IL 61801), and William D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL 61801)

A number of new imaging modalities collect data from cylindrical platforms. *In vivo* imaging needles and intravascular ultrasound imaging catheters are examples of this geometry, where imager rotation and translation parallel to the cylinder axis are the only allowed motions. Efficient three-dimensional ultrasound image formation in these cases can be challenging when the aperture is small and/or highly curved. A frequency-domain imaging algorithm is obtained by approximating the free-space point spread function in cylindrical coordinates and obtaining its Fourier transform by analogy with the equivalent problem in Cartesian coordinates. We further propose an effective use of limited aperture by placing a focused transducer across the aperture, thereby creating a virtual source at the focus which is treated as a real, unfocused source by the imaging algorithm. This approach retains the simplicity and potential angular resolution of a small single element, yet permits full use of the available probe aperture and a higher energy output. Computer simulations and experimental ultrasonic results with wire targets show that this imaging technique attains the theoretical resolution dictated by the operating wavelength and transducer characteristics. [Work supported by NIH CA 079179.]

5:15

**2pBB15. Experimental results of real-time freehand elasticity imaging.** Timothy Hall, Yanning Zhu, Candace Spalding, and Larry Cook (Radiol. Dept., Univ. of Kansas Med. Ctr., 3901 Rainbow Blvd., Kansas City, KS 66160, thall@kumc.edu)

A system for high-speed calculation of tissue elastic properties using a clinical ultrasonic imaging system will be presented. The hypothesis driving this development is that real-time feedback of elasticity images is essential in obtaining high-quality data (consecutive images with high spatial coherence). Extensive experience with laboratory fixtures and off-line processing of elasticity data showed that problems occurring in data acquisition often resulted in poor elasticity image quality. The delay in observing the resulting images, due to off-line processing, resulted in slow progress in developing experimental techniques and signal processing strategies. When high-quality data were obtained, high contrast-to-noise images were available. Initial experience with real-time freehand elasticity imaging shows that images with similar high contrast to noise can be obtained. Results in volunteer patients have shown that high-quality elasticity images are easily obtained *in vivo* for a variety of breast pathologies, and the changes in breast tumor elasticity during the course of chemotherapy can be monitored. Video of real-time elasticity imaging will demonstrate the ease with which these results are obtained. [We are grateful for the support by USAMRAA DAMD17-00-1-0596, Siemens Medical Systems, and NSF BES-9708221.]

## Session 2pEA

## Engineering Acoustics: Analysis and Modeling of Acoustical Systems

Stephen C. Thompson, Chair

Knowles Electronics, 1151 Maplewood Drive, Itasca, Illinois 60143

Chair's Introduction—1:00

## Contributed Papers

1:05

**2pEA1. Numerical model of acoustic interactions between class IV flextensional transducers using a modal method.** John B. BlottmanIII (Naval Undersea Warfare Ctr., Newport, RI 02841, blottmanjb@npt.nuwc.navy.mil)

Acoustic projectors assembled in an array experience an interaction effect as a result of the coupling of their individual radiated powers through the acoustic medium. The acoustic interaction may be expressed as a superposition of modal-mutual radiation impedances. A hybrid modeling technique is presented that combines the boundary integral method with a piece-parts equivalent circuit. The circuit consists of assembling a motional branch for the piezoelectric driver and for a set of flexural modes of the flextensional shell. The *in vacuo* eigenmodes of the shell are determined using the Atila finite element method. Modal radiation impedances are generated using the Chief boundary integral equation method. The resulting modal-mutual impedances are applied to the motional branches through a set of source-coupling terms. The circuit then provides a set of modal participation factors. Results compare well to measurements and to full finite element models of small close-packed arrays. [Work supported by ONR and NUWC ILIR.]

1:20

**2pEA2. Extending nearfield acoustical holography past intermediate sources.** Edward Zechmann and J. Adin MannIII (Aerosp. Eng. and Eng. Mech., Iowa State Univ., 2271 Howe Hall, Ames, IA 50011)

With planar nearfield acoustical holography (NAH), the complex acoustic pressure and the acoustic velocity vector can be estimated on a plane near a complex source surface to identify individual noise sources. However, there are often intermediate sources between the measurement plane and the primary source plane. The NAH technique in its classical form cannot be extended past the intermediate source. A method will be presented to remove the intermediate sources so that NAH can be extended to the surface of the primary source. The normal velocity at the surface of an intermediate source is used to estimate the complex pressure due to the intermediate source on the measurement plane, which is then subtracted from the measured data, providing an estimate of the measured sound pressure if the intermediate source was not present. NAH is then used to project to the surface of the primary source. The source removal process was tested with simulated and measured data. Results for two methods to estimate the pressure caused by the intermediate source, inverse NAH and a point source approximation, will be presented. The results show the effectiveness and limitations of the source removal process. [Work supported by Ford Motor Company.]

1:35

**2pEA3. A trimodal directional transducer.** Alexander L. Butler, John L. Butler (Image Acoustics, Inc., 97 Elm St., Cohasset, MA 02025), and Joe Rice (Naval Postgrad. School, Monterey and SSC, San Diego, CA)

A cylindrical transducer, which achieves a directional horizontal beam through the addition of the first three extensional modes of radial vibration is presented. The three modes are the omni, dipole, and quadrupole modes, related in resonant frequency by the formula  $f_n = f_0(1 + n)^{1/2}$ , where  $n=0$ ,

1, and 2, respectively. The addition of the quadrupole mode to the omni and dipole modes yields a frequency independent horizontal beam that has greater directionality than the commonly used two-mode scheme, first described by S. L. Ehrlich and P. D. Frelich [U. S. Patent 3,290,646 (1966)]. Theoretical and measured beam pattern results on two different trimodal cylindrical transducer designs are presented and compared. One of the transducers, fabricated from rings of the same diameter, operates in the vicinity of 20 kHz and combines modes with three different resonant frequencies. The second transducer, fabricated from rings of different diameters, operates in the vicinity of 10 kHz and combines modes with the same resonant frequency. It is shown that steered beamwidths of 90° and front to back ratios of 10 dB can be readily obtained from comparatively small diameter transducers. [SBIR work supported by ONR and SPAWAR.]

1:50

**2pEA4. Development and validation of a computational process for pass-by noise simulation.** Zhidong Zhang, Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, 2600 Draper Rd., Ann Arbor, MI 48109-2145), T. Allen, and K. Y. Zhang (Ford Motor Co.)

The indirect boundary-element analysis is employed for developing a computational pass-by noise simulation capability. An acoustic field reconstruction process is developed in order to generate the definition of the main pass-by noise sources in a computational model. Numerical boundary-element models that characterize the individual sources are combined in order to develop a system model for the pass-by noise simulation. The acoustic field reconstruction process is validated initially by employing analytical solutions. The numerical techniques are also validated through comparison between numerical results and test data for component-level and system-level analyses. Specially, the source definition capability is validated by comparing the actual and the computationally reconstructed acoustic field for an engine intake manifold. The overall pass-by noise simulation capability is validated by computing the maximum overall sound-pressure level for a vehicle under two different driving conditions. [Research supported by the Ford University Research Program.]

2:05

**2pEA5. Development of a model for acoustic liquid manipulation created by a phased array.** LeAnn E. Faidley and J. Adin MannIII (Iowa State Univ., 2271 Howe Hall, Ames, IA 50010, lfaidley@iastate.edu)

Acoustic Liquid Manipulation (ALM) is the application of the nonlinear effects of high-powered ultrasound to move buoyant objects, create fluid flow, or manipulate fluid-fluid interfaces. Work being done at NASA Glen Research Center indicates that this technique has the possibility to be used to control the location of air bubbles in space vehicle fuel tanks, keeping them away from outlet lines. In order to determine whether ALM is feasible for this and many other possible applications, a model that

predicts the nonlinear behavior of sound created by a variety of source configurations needs to be developed. This was the goal of this project. In this paper the development of a model for radiation pressure, acoustic streaming, and acoustic heating due to a phased array configuration will be outlined. The results of the model will be discussed, and a comparison will be made with experimental data.

2:20

**2pEA6. Optimization of micro-loudspeaker radiation via finite element modeling.** Anthony R. Bontomase and Courtney B. Burroughs (Grad. Prog. in Acoust., The Pennsylvania State Univ., State College, PA 16802)

Measurements and predictions from numerical models for the response of and radiation from a micro-loudspeaker are compared. The numerical models are then used to examine various means for improving the performance of the loudspeaker radiation by varying the material properties of the loudspeaker. A commercial finite element code uses measured material properties to predict the diaphragm velocity distribution of lower order modes. Noncontact laser vibrometer measurements on the excited speaker are compared with the FEM predictions. The radiation from the loudspeaker, at frequencies of resonance, is predicted by a boundary element model. Diaphragm material properties (density, thickness, and elastic modulus) were varied and the sensitivity of the radiation to the material properties of the diaphragm determined.

2:35

**2pEA7. Modeling of perforated absorbing silencers.** Ahmet Selamet, Iljae Lee (The Ohio State Univ., 930 Kinnear Rd., Columbus, OH 43212, selamet.1@osu.edu), and Norman T. Huff (Owens Corning, Inc., Granville, OH 43023)

The acoustic characteristics of single-pass perforated ducts filled with absorbing material are studied experimentally and analytically. The transmission loss predictions from three models [lumped, one-dimensional decoupled, and three-dimensional boundary element method (BEM)] are compared with experimental data from an extended impedance tube setup in the absence of mean flow. Experimentally determined complex characteristic impedance and propagation constant are used to account for the wave propagation through the absorbing material. The perforation impedance suggested by Sullivan [J. Acoust. Soc. Am. **64**, 207–215 (1978)] is modified to incorporate the effect of absorbing material. The results show that: (1) the effect of absorbing material on the perforation impedance needs to be taken into account for accurate predictions; (2) silencers filled with absorbing material exhibit significantly higher attenuation at higher frequencies than those without the filling material; (3) the three-dimensional BEM shows good agreement for the overall frequency range of interest (0–3200 Hz), while the one-dimensional approach is reasonable at relatively low frequencies; and (4) the lumped model, which treats the filled silencer as a Helmholtz resonator, may provide an approximate frequency for the location of peak attenuation.

2:50–3:05 Break

3:05

**2pEA8. Interior noise prediction using a broadband high-frequency boundary-element method based on energy and intensity variables.** Linda Franzoni, Brett Lussier, and Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708-0300)

The analysis of high-frequency broadband sound fields is computationally intensive if it is done on a frequency-by-frequency basis. In addition, traditional numerical methods require that the boundary-element or finite-element model be discretized into elements that are fractions of a wave-

length in size. Despite the high-resolution numerics, in many cases the resulting broadband mean-square pressures are spatially fairly uniform. In this work a boundary element approach has been developed that uses uncorrelated broadband energy sources whose strength and directional properties satisfy certain conditions at the boundary. This method considers all frequencies at once and the element size is large compared to a typical wavelength in the frequency band. There is good agreement between the energy-based boundary-element method and exact solutions for a three-dimensional enclosure with an interior sound source. Spatial correlation effects, which are not in the current model, are also discussed. [Work sponsored by NSF.]

3:20

**2pEA9. Broadband, multimode, free-flooded, baffled circular ring projectors.** Boris Aronov, Tetsuro Oishi, Lawrence Reinhart, and David A. Brown (Acoust. Res. Lab., Elec. Eng., Univ. of Massachusetts–Dartmouth, North Dartmouth, MA 02747 and BTECH, 1445 Wampanoag Tr., S-115, East Providence, RI 02915)

In many underwater applications, it is required to have broadband unidirectional projectors with a small electrical  $Q$  and large electrical power factor. Cylindrical transducers made of circular piezoceramic rings with a part of the surface baffled can be employed to achieve unidirectionality in the horizontal plane over a broad frequency range. In our presentation, it is shown theoretically and experimentally that the operational frequency range of a projector can be extended by simultaneously exciting the 0 (breathing) and 1 (dipole) modes of ring vibration. In addition, the corresponding resonance modes of the internal volume of the free-flooded transducer can further enhance the frequency response. The results of an experimental investigation of the electroacoustical parameters of multimode fluid-filled cylindrical transducers are presented. [This work was supported in part by SBIR N99-011: ONR321SS, ONR36-(J.Rice), BTECH(IRaD), and ONR321SS(Lindberg).]

3:35

**2pEA10. Modeling absorbent finite element for numerical dissipation.** M. A. Picard Lopez, P. E. Solana Quiros, and J. V. Arizo Serrulla (Univ. Politecnica de Valencia, E.T.S. Ing. Industriales, E.T.S. Ing. Caminos Canales y Puertos, Camino de Vera S/N, 46022 Valencia, Spain)

The modeling of the absorbent materials is of great importance for accurate results of the sound field in irregular enclosures using different models by means of finite elements. In this paper different models of porous materials are proven, some already well known due to Craggs, for different geometries of enclosures. The computed results are compared with experimental results obtained demonstrating the reliability of the method and their precision, as well as the validity of the modelization for the absorbent elements by means of their placement in different positions inside the enclosure and for different frequencies. The physics properties used by the absorbent materials are the porosity, the mass density, the flow resistance, and the structure factor. The kinetics, potential, and dissipation energy were considered by the pattern variational used in the FEM. For the discrete distribution of points of the enclosure and the results obtained, averaging of the average residuals has been made. On the other hand, between computed and measured sound pressure values, a Pearson's correlation coefficients has been carried out. There are about 0.8 in numerous cases, an intermediate frequencies. With the absorbents and the numeric method employed, good approaches have been reached.



**2pEA11. Study on acoustic transmission loss characteristics using artificial neural network approach.** A. Ramachandraiah (Civil Eng. Dept., IIT Madras, Chennai-36, India) and G. Maheswari (IIT Madras, Chennai-36, India)

The sound transmission losses of multilayer walls are determined by the physical properties of the component materials. Various analytical methods exist to evaluate the transmission loss of sound in materials. Each method has its own limitations resulting in differences of the predicted and the experimental values. Of late the neural networks have emerged as powerful pattern recognition techniques. In this approach the network learns the similarities among the patterns and infers solutions from the complex, distorted data, which generally the conventional approaches fail to do. Transmission loss of materials in building acoustics seems to be one field where the neural network can be addressed. The transmission loss characteristics of a partition are generally governed by factors like density, thickness of partition, presence of absorptive material, etc. The relation between the input and output parameters is a complex one and may at times be difficult to determine by conventional methods. A neural network has been developed to predict the transmission loss and transmission class values of composite partitions with reasonable precision. The generalization of the network is also tested. The performance of the neural network is observed and the results obtained through neural network approach are analyzed and discussed.

**2pEA12. Simulating an audio beam: A second-generation wavelet collocation method to numerically solve the Khokhlov-Zabalotskaya-Kuznetsov equation.** Kelvin C. M. Lee (DSP Lab., School of Elec. and Electron. Eng., Nanyang Tech. Univ., Nanyang Ave. S., 639798 Singapore), Yew-Hin Liew, and Woon-Seng Gan (ecmlee@ntu.edu.sg)

The Khokhlov-Zabalotskaya-Kuznetsov (KZK) equation is a nonlinear parabolic wave equation which describes an intense finite-amplitude directional sound beam and accounts for the combined effects of diffraction, absorption, and the nonlinearity behavior of the sound beam in air. A general exact solution has not been found for the KZK equation. Nevertheless, it is usually solved by approximate methods or numerical techniques. A second-generation wavelet collocation method using a lifting scheme is used to numerically solve the KZK equation in a highly accurate and efficient manner and has been found to have significant improvements over the conventional finite-difference scheme. Using a new approach to construct wavelets on the interval in the spatial domain (independent of the Fourier transform), the method uses a computational grid which adapts dynamically in time to allow for solution refinement in local regions with sharp transitions, e.g., development of shocks. It is also found to be particularly effective in the treatment of nonlinear terms and general boundary conditions in the equation. Furthermore with a lifting scheme, the wavelet transform can be computed in place of the original signal without the need for auxiliary memory.

TUESDAY AFTERNOON, 5 JUNE 2001

SALON FOYER, 1:00 TO 2:00 P.M.

### Session 2pEDa

#### Education in Acoustics: Undergraduate Research Poster Session

E. Carr Everbach, Chair

*Department of Engineering, Swarthmore College, 500 College Avenue, Swarthmore, Pennsylvania 19081-1397*

#### Contributed Papers

All posters from Session 2aEDb will be on display from 1:00 p.m. to 2:00 p.m. Authors of even numbered papers will be at their posters from 1:00 p.m. to 2:00 p.m.

TUESDAY AFTERNOON, 5 JUNE 2001

SALONS VII/VIII, 2:00 TO 4:00 P.M.

### Session 2pEDb

#### Education in Acoustics: Take Fives for High School Presentations: Sharing Ideas for Teaching Acoustics

Victor W. Sparrow, Chair

*Graduate Program in Acoustics, Pennsylvania State University, 157 Hammond Building, University Park, Pennsylvania 16802*

#### Chair's Introduction—2:00

Do you have a novel demonstration, a new laboratory experiment, a favorite video, a recorded sound example, or a new idea for teaching acoustics which you are willing to share with your colleagues? At this session a sign-up board will be provided for scheduling presentations. No abstracts are printed. Presenters are encouraged to have handouts to distribute. Multiple presentations are acceptable (not consecutively). Presentations are limited to 5 minutes. Keep them short! Keep them fun!



**Session 2pMU****Musical Acoustics: Experimental Musical Instruments**

James P. Cottingham, Chair  
*Department of Physics, Coe College, Cedar Rapids, Iowa 52402*

**Chair's Introduction—1:00*****Invited Papers*****1:05**

**2pMU1. Contemporary ideas in musical instrument making: An overview of recent trends.** Bart Hopkin (Exp. Musical Instruments, P.O. Box 784, Nicasio, CA 94946)

In this talk I will discuss some particularly inventive ideas that have arisen in acoustic musical instrument making in recent years. We'll look at broad trends in instrument design as well as specific instrument types. We will cover acoustic chorusing instruments such as Sharon Rowell's Huaca and Richard Water's Waterphone, musical uses of longitudinally vibrating strings such as Ellen Fullman's Long String Instrument, just intonation and harmonics-oriented instruments such as Hans Reichel's Pick-Behind-the-Bridge Guitar and Jacques Dudon's Photosonic Synthesizer, balloon-mounted and bowed-metal instruments from the Baschet brothers and Tom Nunn, and many more. There will be sound recordings and visuals, and I may be able to bring one or two smaller instruments along for show and tell.

**1:35**

**2pMU2. New and unusual percussion instruments.** Thomas D. Rossing (Phys. Dept., Northern Illinois Univ., DeKalb, IL 60115)

Many new and unusual percussion instruments have been developed in recent years, and more are in the experimental stage. What is often termed "contemporary sound" makes extensive use of percussion instruments. We will describe the acoustical properties of a number of new percussion instruments made of wood, metal, glass, and stone. Normal modes of vibration and sound radiation will be emphasized.

**2:05**

**2pMU3. Sounding clay: Pre-Hispanic flutes.** Susan Rawcliffe (P.O. Box 924, San Pedro, CA 90037)

Ms. Rawcliffe makes and plays ceramic flutes and sound sculptures. Many were inspired by her explorations into ancient and wonderful wind instruments. To build a better flute, she studied construction methods of both contemporary and pre-Hispanic artists. The laws of acoustics dictate the possibilities for instrument construction within which design decisions are made according to cultural and individual preferences. She makes acoustical copies, learns to play them, then reinvests her insights, evolving through stages into new instruments, which inspire with their wonderful scales and evocative timbres. For 30 centuries, societies from the Olmecs to the Mayans and Aztecs developed a unique flute organology in a great diversity of form, timbre, and tunings. Many of the most complex innovations of the pre-Hispanic ceramists resulted in instruments of restricted pitch but rich timbre. They are not so much the "remains of a bygone art, as the sacred sound symbols of a now vanished cult." In reviving this lost craft and science, Ms. Rawcliffe is able to play sounds that may not have been heard for 1000 years. For this presentation, using slides of her own and of ancient flutes, and performances on her flutes, she will discuss and illustrate the above issues.

**2:35**

**2pMU4. The development of "21st century" acoustic and orchestral instruments.** Patrick Ozzard-Low (Ctr. for New Musical Instruments, London Guildhall Univ., 41 Commercial Rd., London E1 1LA, UK)

The work of the Centre for New Musical Instruments (CNMI), recently established at London Guildhall University, is reported. In addition, a wide range of original slides is shown, providing an overview of international innovations relevant to the development of new versions of existing acoustic orchestral instruments, especially those offering new forms of expression to composers and performers. In view of the overwhelming majority of developments of electronic as opposed to acoustic instruments during the 20th century, the purpose of CNMI is to encourage innovations of the latter. Acoustic instruments designates not only those traditionally referred to by this term, but also those in which elements of sound generation may be electromechanical or electroacoustic, although sound diffusion is not through a loudspeaker. Three areas of innovation are discussed: (i) purely mechanical instruments; (ii) electromechanical and electroacoustic hybrids; and (iii) instruments designed for alternative tuning systems. All the orchestral families are considered. The mutually reinforcing possibilities and limitations of these elements, together with the evolving aesthetic issues of contemporary and other musics, suggests the importance of specific areas of future research and instrument making. Amongst these is the necessity of interdisciplinary perspectives and collaboration.

3:15

**2pMU5. The Woodstock Gamelan.** Lydia Ayers and Andrew Horner (CS Dept., HKUST, Clear Water Bay, Kowloon, Hong Kong)

This paper considers the spectral properties of the Woodstock Gamelan, a 3-octave set of tubular chimes built by Woodstock Percussion in upstate New York. One of the main features of the instrument is expandability, and it includes 75 microtones in the middle octave. The justly tuned instruments of Harry Partch inspired the instrument. The Woodstock Gamelan has two types of aluminum tubes. Racks support the tubes down to Eb4, and these have a vibraphonelike timbre. The larger hanging tubes go down to G3, and have larger diameters than the rack tubes. The sound of the hanging tubes is similar to that of orchestral chimes. The Woodstock Gamelan has five exponentially decaying partials, and their frequency ratios are the same in the low and high registers. The frequency ratios measured were about 1, 2.69, 5.15, 8.38, and 12.08. The ratios are close to the just ratios of 1, 2.667, 5.333, 8.533, and 12. A Csound model for the Woodstock Gamelan has been developed. Listening tests show that the model produces tones nearly indistinguishable from the original. The model produces attractive related timbres by simple changes to the parameters.

3:45

**2pMU6. New woodwind instruments.** Lewis Jones (Musical Instrument Technol., London Guildhall Univ., 41 Commercial Rd., London E1 1LA, UK, [lioness@lgu.ac.uk](mailto:lioness@lgu.ac.uk))

Progress towards the development of woodwind instruments designed to play fluently in tuning systems other than 12-note equal temperament is described. The problems of closing many toneholes with only ten digits are outlined, and past solutions, depending chiefly on rod and lever keywork, are briefly reviewed. The alternative of largely keyless toneholes, as embodied in Jones and Armitage's 19ET recorders (1999) is considered. The electromechanical means of tonehole closure introduced in Giles Brindley's "logical bassoon" is taken as a point of departure for a range of woodwind instruments including flutes dividing the octave into as many as 36 steps, others of exceptionally deep pitch, and others, narrow but of great length which, by exploiting the higher harmonics of the air column can, without impossibly many toneholes, produce divisions of the octave into 72 or 96 steps. Software mediates between practicable fingerings and at times complex patterns of tonehole movement. A scheme of fingering is proposed which is intended to enable a single player to learn to play in many alternative tuning systems while preserving as many essential parallels between fingering and sound as possible.

4:15

**2pMU7. Conventional harmonic behavior associated with the equal tunings of 17 and 19 notes.** Easley Blackwood (5300 S. Shore Dr., Chicago, IL 60615, [ebblackwood@interaccess.com](mailto:ebblackwood@interaccess.com))

It will be demonstrated how diatonic behavior associated with the equal tunings of 17 and 19 notes is virtually identical to that of the standard 12-note equal tuning. However, levels of discordance are noticeably different among the three tunings. The reasons for this will be shown in detail. It will also be shown how the conventional musical notation—the five-line staff with treble and bass clef, as well as accidentals—is an accurate and comprehensible representation of the tunings of 17 and 19 notes. Musical illustrations in the form of brief (4-minute) compositions, electronically created, can be provided. Funding for this study was provided by a grant from the NEH, and administered through Webster College, St. Louis.

4:45–5:45

Informal Performance and Demonstration of Experimental Instruments

2p TUE. PM

**Session 2pNS****Noise and Architectural Acoustics: Motor Vehicle Interior Noise Control**

Daniel R. Raichel, Chair

2727 Moore Lane, Fort Collins, Colorado 80526

**Chair's Introduction—1:00*****Invited Papers*****1:05****2pNS1. Customizing vehicle interior noise to reflect the brand image.** Hans P. Schedl (Audi AG, 85045 Ingolstadt, Germany)

Noise control engineers in their endeavors to make vehicles as quiet as possible have succeeded thanks to major strides in measurement techniques. The different noise sources in vehicles are now quite well known. Additionally, sounds may be simulated and easily modified on simple PCs. Through psychoacoustic means, we are able to evaluate and understand the effects of noise on human beings. A totally new approach to vehicle interior noise now exists. Engine noise conveys meaningful information to the driver, and thus, noise characteristics constitute a design attribute. Manufacturers are increasingly establishing design rules for the sound of their cars to meet special requirements and preferences of their potential customers. The design rules are meant to yield a vehicle brand-specific sound. This paper analyzes the positive and negative effects of noise on both driver and passengers in a vehicular environment and how sound is used as a marketing and public relations instrument. We also examine briefly the way sound-effect design affects a project's progress, cost and time limits. An example will demonstrate how the interior noise of a basic car is modified to either sound like an upper-class luxury sedan or like a high-performance sports car.

**1:30****2pNS2. Quantifying customer perceptions of impulsive vehicle noise.** Mike Blommer, Scott Amman, and Deanna Hoffman (Ford Motor Co., SRL Bldg., MD 2115, P.O. Box 2053, Dearborn, MI 48121, mblommer@ford.com)

Sounds such as spark and diesel knock, squeaks and rattles due to body and suspension components, gear rattle, and other impulsive events, can occur in vehicles and be a major source of customer dissatisfaction. It is desirable to not only know the detection thresholds of these impulsive events, but also the relative annoyance they impart on the customer once they become audible. This work describes research addressing both aspects. The first part of the paper presents a generalized detection model of impulsive events in common vehicle background noises (e.g., wind, road, and powertrain noise). Important properties of the model are the combination of impulsive event information across frequency and also the effect of overall background noise level on detection thresholds. Application of the model to predict detection thresholds for spark knock and also squeaks and rattles is presented. Comparisons are made to measured subjective thresholds. The second part of the paper presents research in objectively quantifying the relative annoyance of impulsive sounds as a function of their temporal and amplitude distributions, as well as their loudness. Comparisons are made to objective metrics used for characterizing single impulsive events.

**1:55****2pNS3. Prediction of vehicle passenger compartment noise due to turbulent flow excitation.** Sean Wu and Guoming Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

An engineering computer model is developed for estimating noise transmission into a vehicle passenger compartment, given the characteristic dimensions of the vehicle and turbulent flow excitation. In this model, the effects of window offset and leakage, and those of power spectral densities due to separated and reattached flows are taken into account. Empirical formulations are developed to estimate the reattachment line and the power spectral densities of separated and attached flows. The effect of leakage is accounted for by a correction factor. The computer model thus developed is validated on a greenhouse vehicle model. Experiments are conducted in a wind tunnel at the Chrysler Technology Center. Pressure fluctuations due to separated and attached flows on the side window are measured and compared with the calculated values. These pressure fluctuations are taken as input to a computer program, called VibroAcoustic Payload Environment Prediction System based on Statistical Energy Analysis, to calculate the noise spectrum, given the A-pillar angle, window size and offset, vehicle yaw angle, and wind speed. The predicted noise spectra inside the passenger compartment are compared with the measured ones under various flow speeds. Satisfactory agreements are obtained in all cases. [Work supported by a Chrysler Challenge Fund.]

**2:20****2pNS4. Future trends in the automotive sound package industry.** Pranab Saha (Kolano and Saha Engineering, Inc., 3559 Sashabaw Rd., Waterford, MI 48329, prsaha@kandse.com)

With various challenges in the automotive industry, there are enormous opportunities in sound package treatments. Although the fundamentals of the sound package practices have changed very little, the technology has changed significantly, and this has brought about new opportunities. More and more the supplier companies are asked to join OEM companies in actively assisting with the

optimization of sound package treatments in vehicles. This environment encourages innovation and suppliers now can bring in new concepts and products that were not thought of before. This adds more value which is a combination of performance, cost, and weight. New products that are coming to the industry are products like fine fibers and barriers for airborne noise, and sprayable damping materials for structure-borne noise. In addition, there are new test methods that are intended to validate these products which will help acoustics in the vehicle. This paper discusses these trends, and associated opportunities and challenges in the automotive sound package industry in the years to come.

2:45–2:55 Break

2:55

**2pNS5. Computational methods for prediction and control of automotive interior noise.** S. T. Raveendra (Collins & Aikman, 47785 W. Anchor Court, Plymouth, MI 48170)

Prediction of noise and vibration characteristics is important to the automotive community. Typically, noise and vibration testing are performed on prototypes of vehicles during pre-production stages of the development. Although testing is an essential tool, it is not the most effective tool since it is done at the later stages of the development. As a result it is not possible to make major design modifications that may be necessary for the effective treatment of the noise and vibration problems. On the other hand, computational modeling and simulation is feasible at the early stages of the design and is thus effective for the management of noise and vibration problems. This presentation will demonstrate how boundary-element method based techniques can be used to predict and effectively control noise in the interior of an automobile. Initially, a technique based on acoustic contribution analysis that intuitively allows the reduction of noise in the interior of an automobile will be demonstrated. This will be followed by a demonstration of an integrated experimental/computational simulation technique based on the near-field acoustical holography that permits the identification of panel vibrations that contribute to the noise in the interior of an automobile.

3:20

**2pNS6. Effects of brake pressure and temperature on squeal noise.** Alessandro Mattiuzi Balvedi (MSC Laminates and Composites, Inc. and Univ. of Santa Catarina, Santa Catarina, Brasil) and Ahid Nashif (MSC Laminates and Composites, Elk Grove Village, IL 60007-5995)

Brake squeal noise has been an important issue for the automotive industry. This phenomenon is related to the modal coupling of the rotor and the pads of a brake system. When a mode of the pad matches with a mode of the rotor, the system becomes unstable. The consequence is the generation of additional vibration energy that cannot be dissipated by the system. In this manner, a solution is to improve the damping capacity of the brake system. This is accomplished by use of metal and viscoelastic materials laminates, where some of the vibrational energy is dissipated by shearing or extending the viscoelastic layer. Since the dynamic behavior of brake systems is a function of frequency, temperature, and brake pressure (friction force), it is of utmost importance to verify the influence of such parameters in order to design and select a proper multilayer treatment. This work presents a procedure to verify the influence of these parameters in the coupling of rotor and pads modes. Different conditions of pressure and temperature are applied on a brake system and the dynamic of the system related to squeal noise is evaluated.

### Contributed Papers

3:45

**2pNS7. Burst mode dither control of automotive brake squeal.** Mawuli Dzirasa, Kenneth A. Cunefare (The Georgia Inst. of Technol., Atlanta, GA 30332-0405), Victor Rastelli, and Nila Montbrun (Universidad Simon Bolivar, Caracas 108, Venezuela)

Brake squeal is an annoying phenomenon, generally occurring when vehicles brake at slow speeds. Significant effort is expended by the automotive industry to avoid squeal during the development of brake systems. Nonetheless, expensive warranty claims still occur when such efforts are not completely successful. Recent research demonstrated that introducing dither control into a brake is an effective means of suppressing squeal. Dither control is a method by which high-frequency disturbances are introduced into a system. This suppresses the friction induced squeal response at lower frequencies, thus eliminating the audible brake squeal. Dither control, when applied at a 100% duty cycle, is an effective means of suppressing brake squeal. This presentation focuses on the potential to use burst mode dither control for squeal suppression. Burst mode dither control is characterized by duty cycles of less than 100%. Dither is introduced to a brake by placing a piezoelectric stack actuator in the piston of a floating caliper brake. The piezoelectric stack actuator is driven by a 20-kHz burst mode signal, and the impact upon the signal is assessed. Burst mode dither with duty cycles as low as 50% are shown to be as effective in suppressing brake squeal as dithering at a 100% duty cycle.

4:00

**2pNS8. An experimental investigation of techniques for reducing flow-induced cavity resonance.** Paul Zoccola (Carderock Div. Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817, zoccolapj@nswccd.navy.mil)

Excitation of cavity resonance by flow over an aperture is often a source of unwanted noise in aerospace, automotive, and marine applications. An experimental investigation of three resonance reduction techniques was conducted. These were: a fence at the upstream edge, fluid injection, and a new technique developed by the author, in which fluid from the boundary layer is diverted into the cavity. Spectra of the pressure in the cavity were obtained for various flow speeds. Results show that the fence and the boundary-layer diversion technique have the effect of reducing the Strouhal number of the flow-excited sheartones, delaying resonance to higher speeds. Reduction at the resonant speed is also observed. The boundary layer diversion technique was much more effective than the fence. The Strouhal number reduction effect is not observed with fluid injection. The effect on resonance reduction by fluid injection as a function of several parameters, including the rate of fluid injection, is discussed.

**2pNS9. Visualizing acoustic field inside a passenger vehicle compartment.** Moondra Manmohan and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

The objective of this study is to examine the effectiveness of the HELS method [S. F. Wu and J. Yu, *J. Acoust. Am.* **104**, 2054–2060 (1998); S. F. Wu, *ibid.* **107**, 2511–2522 (2000)] in visualizing the areas that are prone to noise transmission into a full-size vehicle passenger compartment due to engine, powertrain system, tires, and wind excitations. The input to the HELS formulation is the acoustic pressure measured inside the compartment. No vehicle geometry and dimensions are necessary. The optimum number of expansion functions is determined by minimizing the sum of the squared errors with respect to the measured data. Once the HELS formulation is established, the acoustic pressure anywhere including the vehicle interior surface can be determined. The normal component of the surface velocity can be reconstructed in a similar manner. Once these quantities are specified, the normal component of the time-averaged acoustic intensity and acoustic energy flow inside a vehicle passenger compartment can be visualized. This three-dimensional acoustic image can provide valuable insight into vehicle interior noise reduction. The

effects of number and locations of measurements on the accuracy of reconstruction are investigated. [Work supported by the Daimler-Chrysler Challenge Fund.]

**2pNS10. Fiber optic sensor system for structural acoustics control.** Miao Yu and Balakumar Balachandran (Dept. of Mech. Eng., Univ. of Maryland, College Park, MD 20742, mmyu@eng.umd.edu)

Recent work on acoustic measurements using a fiber-tip-based Fabry–Perot sensor system is presented. A single Fabry–Perot sensor using a path matched Mach–Zehnder interferometer is developed, and by taking advantage of an integrated optical circuit phase modulator, a digital demodulation scheme based on the phase stepping technique is developed. It has been determined that this system can be used to detect acoustic fields in the frequency range of 20 Hz–6 kHz with a sensitivity of 0.9 rad/Pa. This sensor is designed to be used in a multiplexed architecture to provide inputs to a structural acoustic control system. A series of experiments are performed to investigate the possibility and potential use of this sensor system for acoustic noise detection. In this study, initial test data from the prototype optical sensor microphone are presented and the envisioned multiplexed sensor scheme and control system are illustrated.

TUESDAY AFTERNOON, 5 JUNE 2001

CRYSTAL ROOM, 1:00 TO 3:35 P.M.

## Session 2pPAa

### Physical Acoustics: Extraterrestrial Acoustics

George Mozurkewich, Chair  
10671 Canton Center Road, Plymouth, Michigan 48170

Chair's Introduction—1:00

#### Invited Papers

1:05

**2pPAa1. “Sonoluminescence” from the early Universe.** Michael Turner (Dept. of Astron. & Astrophysics, Univ. of Chicago, 5640 South Ellis Ave., Chicago, IL 60637 and Fermi Natl. Accelerator Lab.)

The tens of microkelvin variations in the temperature of the cosmic microwave background (CMB) across the sky are snapshots of the Universe at 400 000 years of age and a Rosetta Stone for unraveling its earliest history. During the first 400 000 years the lumpy distribution of dark matter excited gravity-driven acoustic oscillations of the ordinary matter. Pressure provided by the photons in the CMB, which were tightly coupled to the ordinary matter, acted as the restoring force for these compressional modes. During an oscillation, photons were alternately compressed and heated, and rarefied and cooled. Ordinary matter and photons decoupled at 400 000 years, and CMB photons streamed freely to us, creating a snapshot of modes caught at maximum compression (hot spots) and maximum rarefaction (cold spots) at this early time. The pattern of hot and cold spots on the CMB sky today is being used to determine the curvature of the Universe, the Hubble constant, the total amount of ordinary matter and of dark matter, and to test theories of the early Universe including inflation.

1:35

**2pPAa2. Helioseismology—sound inside the sun.** Frank Hill (Natl. Solar Observatory, 950 N. Cherry Ave., Tucson, AZ 85719)

In 1960, the solar surface was found to be moving radially with a period of 5 min. The cause of this effect was found to be the trapping of acoustic waves in the solar internal thermal gradient. Since the sound fills a resonant cavity, only specific oscillatory modes are present whose characteristics depend on the plasma properties. Thus, the solar interior can be remotely sensed by observing the trapping region's upper boundary at the surface. This led to the field of helioseismology and produced new information about the solar internal rotation rate, composition, neutrino flux, convection zone depth, and internal changes associated with the solar activity cycle. It has also generated progress in remote sensing techniques (acoustic holography, ring diagrams, inversions), observing facilities (GONG Network, MDI instrument on SOHO), and insights into the coupling of turbulent convection and sound. Presumably, other stars also contain sound and their interiors could thus be probed, but so far no conclusive observations of these vibrations have been obtained. [Work supported by NSF and NASA.]



**2pPAa3. Modeling acousto-gravity waves in the atmosphere of Jupiter.** Joseph F. Lingeitch, Michael D. Collins (Naval Res. Lab., Washington, DC 20375), and Catherine Stamoulis (Dept. of Ocean Eng., MIT, Cambridge, MA 02139)

The modeling of wave propagation in the atmosphere of Jupiter became a topic of great interest after the discovery of the fragments of Comet Shoemaker-Levy 9. Circular fronts were observed expanding from several of the impact sites [Hammel *et al.*, *Science* **267**, 1288–1296 (1995)], but the nature of these features has not yet been explained in terms of a wave model. This is due in part to a lack of data, which has been enhanced by direct measurements from the Galileo probe [Seiff *et al.*, *J. Geophys. Res.* **103**, 22 857–22 889 (1998); Atkinson *et al.*, *ibid.* **103**, 22911–22928 (1998)]. There have also been some recent improvements in propagation modeling using parabolic equation techniques, which can handle acousto-gravity waves [Lingeitch *et al.*, *J. Acoust. Soc. Am.* **105**, 3049–3056 (1999)] and the advective effects of winds [Collins *et al.*, *J. Acoust. Soc. Am.* **97**, 2147–2158 (1995)]. These techniques and their application using Galileo data will be presented. [Work supported by ONR.]

**2pPAa4. Global infrasonic monitoring of large meteoroids.** Douglas O. ReVelle (Los Alamos Natl. Lab., P.O. Box 1663, MS J577, Los Alamos, NM 87545)

The hypersonic meteoroid/bolide-atmosphere interaction can generate numerous phenomena, including strong shocks and infrasonic waves. For a nonfragmenting bolide, the line source, blast wave radius,  $R_0$ , is the product of the Mach number and the body diameter. For bolides reaching continuum flow,  $R_0$  can range from  $\sim 1$  m to many kilometers. To be detected,  $R_0$  must exceed  $\sim 10$  m [at a minimum source energy,  $E_s$ , of  $\sim 4.10(7)$  J]. Beyond  $10 R_0$ , a frequency equal to  $C_s/(2.81 R_0)$  will be generated which will change during propagation due to effects of nonlinearity, absorption, and dispersion. Infrasonic arrays are now being routinely used to detect and locate such large near-Earth objects. Data from the CTBT IMS monitoring system will help refine estimates of the large body influx rate. These arrays have baselines of  $\sim 1$  to 2 km using microbarometers with response from  $\sim 10$  to 0.01 Hz. For ranges from 100 to 14 000 km, such bolides have been observed previously with amplitudes from 0.02 to 16 Pa, with periods from 0.50 to 300 s, with total durations from 0.10 to 25 min and with computed acoustic efficiencies varying from 0.01% to  $\sim 10\%$ . The influx rate will be discussed as well as several recent bolides detected by Los Alamos arrays and by US DoD satellites.

### Contributed Papers

**2pPAa5. Absorption of sound in the Martian atmosphere.** Henry E. Bass and James P. Chambers (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677, chambers@olemiss.edu)

Future missions to the planet Mars might include microphones to listen for sounds in the tenuous Martian atmosphere. The chemical composition of the atmosphere is well established by previous missions and ground based observations. The dominant constituent is  $\text{CO}_2$  with a minor amount of  $\text{N}_2$  and argon and smaller amounts of  $\text{H}_2\text{O}$ . The factors important to compute the absorption of sound in a gas are reasonably well known, the most uncertain being the relaxation time of  $\text{CO}_2$  at the low temperatures encountered (200 K–300 K). By extrapolating higher temperature measurements of relaxation times, analytical expressions have been developed for the absorption due to viscosity and thermal conduction (classical absorption), rotational relaxation, and vibrational relaxation. Calculations are presented for a surface level pressure of 6 millibars (600 Pa). The absorption at mid-audio frequencies (500 Hz) is 0.03 (200 K)–0.1 (300 K)  $\text{Np m}^{-1}$ . This is about 100–500 times greater than for the earth's atmosphere (depending upon relative humidity).

**2pPAa6. The other ocean: probing Europa's interior with natural ambient noise sources.** Aaron M. Thode, Michele Zanolin, Sunwoong Lee, Purnima Ratilal, Josh Wilson, and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

Jupiter's moon, Europa, possesses a surface composed primarily of water ice. Gravity and magnetic data collected by the NASA Galileo Orbiter over the past five years have provided increasing evidence that an ocean exists underneath this layer, and that the combined ice/water layer thickness is 80–170 km, although the ice shell thickness remains unknown. The surface is covered by numerous fractures and ridges, believed to have formed in response to tidal deformations generated by Europa's slightly eccentric 85-hour orbit around Jupiter. A recently published model [Hoppa *et al.*, *Science* **285**, 1899–1902] argues that certain cycloidal fractures must form during a single revolution, and propagate horizontally at an average speed of 3 km/h. Considerations from ice mechanics suggest that these propagating fractures would generate significant acoustic energy in the frequency range 0.5–4 Hz, where low attenuation in the ice/water environment is expected. In this presentation an acoustic sound speed profile for Europa is constructed, and standard ocean acoustic techniques are used to demonstrate how an array of geophones on the icy surface could simultaneously localize discrete events and invert for the ice-layer thickness.

## Session 2pPAb

## Physical Acoustics: Materials Characterization

Stanley A. Cheyne, Chair

Department of Physics and Astronomy, Hampden-Sydney College, Box 158, Hampden-Sydney, Virginia 23943

## Contributed Papers

3:45

**2pPAb1. Correlation of resonance spectra excited by acoustic waves on elongated elastic objects with elastic bulk parameters and object identification.** H. Uberall and M. F. Werby (NRL Code 7180, Stennis Space Center, MS 39529 and Dept. of Phys., The Catholic Univ. of America, Washington, DC 20064)

Acoustic resonance spectra are calculated for scattering of acoustic signals from elongated objects composed of six elastic materials with two aspect ratios of 3 and 6. The incident field is along the axis of symmetry and broadside. A comparison is made of the resonance locations of the six materials (brass, nickel, aluminum, steel, manganese, and tungsten carbide) representing a broad spectrum of elastic materials. For the principal resonances the ratios are seen to correspond exactly to the Rayleigh phase velocities on an evacuated half-space, or alternatively to the shear bulk velocity and are a function of the Poisson ratios of the material. Further, the resonance widths are related inversely to the density of the material and the shear velocity (the mechanical impedance of the shear wave). Time-domain calculations are also carried out and the resonance widths and travel times may be identified with the material properties of the target. Thus, the material properties of such objects including elongation may be distinguished for submerged objects, and this is a useful tool for inverse issues.

4:00

**2pPAb2. Gel formation monitoring by acoustic spectroscopy.** Loïc Martinez, Stéphane Serfaty, Brahim Senouci (Lab. d'Electronique Appliquée, Univ. de Cergy, 5 mail Gay-Lussac, F 95 031 Neuville sur Oise Cedex, France, loic.martinez@iupge.u-cergy.fr), Pascal Giesmar (Lab. de Physico-Chimie des Matériaux Organiques, Univ. de Cergy Pontoise, F 95 031 Neuville sur Oise Cedex, France), and Marcel Gindre (Lab. d'Imagerie Paramétrique, UMR CNRS 7623, F 75006 Paris, France)

An acoustic technique in the audible range has been developed to characterize the sol-gel process. Resonances appear at the sol to gel transition of a sol-gel matrix when submitted to an acoustic wave. The range of the associated resonance frequencies leads to a very low propagation speed of sound (about 20 m/s). The resonance frequencies versus time curves, corresponding to the harmonic propagation modes, converge to a unique intersection point with the time axis corresponding to the gelation time  $t_g$ . The temporal evolution of the resonance frequencies features the formation of the network. Actually, the evolution of the matrix is independent of the initial conditions (precursor concentration, hydrolysis rate). Depicting the "reduced frequency"  $f_i/f_i(\infty)$  [ $f_i(\infty)$  is the long-term resonance frequency for the harmonic mode  $i$ ] versus the "reduced time"  $t/t_g$  for various Si concentrations and hydrolysis rates results in a unique curve, revealing the insensitivity of the matrix formation process to the input parameters. A propagation model of an acoustic wave in a cylindrical cavity is proposed, to compute the resonance frequencies as a function of the longitudinal to transverse propagation velocity ratio  $C_L/C_T$ . The comparison with the experimental data is conclusive.

4:15

**2pPAb3. The influence of a strong acoustic cw wave on a weak signal in rock.** Irina Soustova, Veniamin Nazarov, Alexander Sutin, and Andrey Radostin (Inst. of Appl. Phys. RAN, Nizhny Novgorod, Russia)

We are conducting studies of the effect of a strong pump wave on a probe wave in rock. Specifically, we are measuring resonance frequencies and quality factors (applying linear resonant ultrasound spectroscopy) for a relatively weak probe wave in the presence of a strong, cw pump wave. Experiments conducted in sandstone demonstrate a shift of the resonance frequency and increasing of  $Q$  factor in the presence of the low frequency pump wave. The model describing this phenomenon is based on including small hysteretic loops in the stress-strain dependence for the probe wave. The hysteretic loop parameters depend on the loop position on the stress-strain dependence for the pump wave. [Work was partially supported by International Science Technical Center, Grant No. 1369.]

4:30

**2pPAb4. Resonant ultrasound spectroscopy studies of clathrate thermoelectrics.** Michael McGuire, Alem Teklu, Veerle Keppens (Natl. Ctr. for Physical Acoust. and Dept. of Phys., The Univ. of Mississippi, Oxford, MS 38677), David Mandrus, and Brian Sales (Solid State Div., Oak Ridge Natl. Lab., Oak Ridge, TN 37831)

Resonant ultrasound spectroscopy (RUS) measurements have been carried out for 2 cubic clathrate materials,  $\text{Eu}_8\text{Ga}_{16}\text{Ge}_{30}$  and  $\text{Ba}_8\text{Ga}_{16}\text{Ge}_{30}$ . In these materials, the Eu and Ba ions rattle in oversized atomic cages. They have attracted attention as promising thermoelectric materials, having thermal conductivities with a glasslike order of magnitude while maintaining crystalline electronic properties. RUS has proven to be useful for the study of similar cage-like materials [Keppens *et al.*, Nature **395**, 876–878 (1988)], identifying 2 local modes in the filled skutterudite  $\text{La}_{0.75}\text{Fe}_3\text{CoSb}_{12}$ . The RUS measurements we present here were carried out as a function of temperature (2–300K) on single crystals, and allow the determination of the 3 elastic moduli. The results are compared to ultrasonic attenuation measurements on  $\text{Sr}_8\text{Ga}_{16}\text{Ge}_{30}$  and RUS measurements on filled and unfilled skutterudites. [Work supported in part by ONR. Oak Ridge National Laboratory is managed by UT-Battelle, LLC, for the U.S. Department of Energy under Contract No. DE-AC05-00OR22725.]

4:45

**2pPAb5. Diffuse backscattered ultrasound for material characterization.** John Burkhardt (Mech. Eng. Dept., U.S. Naval Acad., Annapolis, MD)

An ultrasonic technique is proposed that allows for the characterization of scattering materials by the Legendre moments of their differential scattering cross section per unit volume. Estimates of Legendre moments are determined from an ultrasonic backscatter theory that models acoustic propagation in the medium as a time-dependent diffusion process. Monte Carlo simulations are presented which support the proposed technique. Initial experimental progress will be reported.

## Session 2pPP

## Psychological and Physiological Acoustics: Physiology; Binaural Processing (Poster Session)

William P. Shofner, Chair

Parmly Hearing Institute, Loyola University of Chicago, 6525 North Sheridan Road, Chicago, Illinois 60626

## Contributed Papers

All posters will be on display from 1:30 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:15 p.m. and contributors of even-numbered papers will be at their posters from 3:15 p.m. to 5:00 p.m.

**2pPP1. Fluid volume displacement at the stapes footplate and round window due to air and bone conduction stimulation.** Stefan Stenfelt (Dept. of Signals and Systems, Chalmers Univ. of Technol., Goteborg, Sweden), Naohito Hato, and Richard L. Goode (Stanford Univ. Medical Ctr., Stanford, CA)

The fluid volume displacement of the two major in- and outlets of the cochlea, the stapes footplate and the round window, are normally considered equal but with opposite phase. However, other channels, such as the cochlear and vestibular aqueducts, may affect the fluid flow in the cochlea. To test this hypothesis, the volume displacement at the stapes footplate and the round window was measured with a laser Doppler vibrometer with air conduction stimulation in seven fresh human temporal bones and with bone conduction stimulation in eight temporal bones. The volume displacement was computed by measuring at five positions on the footplate and by scanning the motion of the round-window membrane at approximately thirty positions. With air conduction stimulation, the average volume displacement of the two windows was within 5 dB of each other and the phase difference was close to 180 deg within the frequency range 0.1–10 kHz. With bone conduction stimulation and below 1.5 kHz the volume displacement of the round window was about 10 dB greater than at the footplate with a phase difference of 150–200 deg. Above 1.5 kHz this difference rolled off with 6 dB/octave and 100 deg/octave.

**2pPP2. Oxidative DNA damage associated with intense noise exposure in a rat.** Luann E. van Campen, William J. Murphy (Hearing Loss Prevention Section, NIOSH, 4676 Columbia Pkwy., M.S. C-27, Cincinnati, OH 45226-1998), and Mark A. Toraason (NIOSH, Hearing Loss Prevention Section, Cincinnati, OH 45226-1998)

Increasing evidence suggests that noise-induced hearing loss may be prevented with antioxidant therapy. In order to appreciate treatment constraints, a biochemical marker(s) of reactive oxygen species (ROS)-induced damage is necessary. Without a marker, the timing of damage and biologically effective exposure(s) cannot be understood fully. This study examined the time course of ROS damage following noise exposure resulting in permanent threshold shift in a rat. Cochlea, brain, liver, serum and urine were analyzed. Oxidative DNA damage was assessed by measuring 8-hydroxy-2'-deoxyguanosine (8OHdG) by high performance liquid chromatography with electrochemical detection (HPLC-EC). Lipid peroxidation was measured via the thiobarbituric acid reactive substance's (TBARS's) colorimetric assay for detection of aldehydes (e.g., malondialdehyde). Auditory brainstem response and distortion product otoacoustic emissions thresholds showed progressive elevation up to 3–8-h post-exposure, then notable recovery at 72 h, and some worsening at 672 h. 8OHdG/dG was significantly elevated in cochlea at 8-h post-exposure, and in the brain and liver at 72 h. TBARS were significantly elevated in serum

at 72 h. This is the first evidence that oxidative DNA damage is present in cochlea following intense noise, and that the timing of damage corresponds to the timing of functional impairment.

**2pPP3. Temporary threshold shift in ABRs and DPOAEs following noise exposure in Long-Evans rats.** William J. Murphy and Luann E. van Campen (NIOSH, Hearing Loss Prevention Sect., 4676 Columbia Pkwy., Cincinnati, OH 45226-1998)

In conjunction with a study of reactive oxygen species induced damage in Long-Evans rats [van Campen *et al.*, ARO abstracts (2001)], the progressions of temporary shifts in auditory brainstem response thresholds and distortion product otoacoustic emission input/output functions were analyzed. Thirty-five rats were exposed to 2 h of 120-dB SPL band-limited noise (7.5–15 kHz). ABR and DPOAE measurements were collected at 1-, 3-, 8-, 72-, and 672-h post-exposure for seven rats per condition. One sham-exposed rat was included in each exposure group. Thresholds for ABR elicited by tone bursts (2, 4, 8, 16, 32 kHz) exhibited the greatest threshold shifts for the 8-, 16- and 32-kHz stimuli. Recovery following noise exposure was greatest for the 72-h group with a lesser recovery for the 672-h group. The DPOAE thresholds (level of the  $f_2$  primary stimulus where LDPOAE was 5 dB above the noise floor) exhibited no significant difference for the 72- and 672-h groups. However, the input/output functions for the 72 group exhibited more DPOAE energy than the 672-h group.

**2pPP4. Measurement and interpretation of stimulus frequency otoacoustic emissions: Input/output functions and transient analyses.** Douglas H. Keefe, Kim S. Schairer, Fei Zhao, and Jeffrey L. Simmons (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131, keefe@boystown.org)

A stimulus frequency otoacoustic emission (SFOAE) is a cochlear-based signal measured in the ear canal in response to sine-tone stimulation. SFOAE responses are potentially related to other auditory-system responses to sine-tone stimulation, e.g., basilar membrane input/output (I/O) functions, and behavioral studies of nonlinearity. In preliminary experiments, system distortion has been measured in an ear simulator using SFOAE and DPOAE paradigms, and different probes. SFOAE I/O functions have been measured in subjects using the double-evoked method [D. H. Keefe, J. Acoust. Soc. Am. **103**, 3489–3498 (1998)] in two approaches: equal-level stimuli in each loudspeaker, or a fixed high level in one loudspeaker and varying level in the other. The latter approach is particularly helpful, because it controls for mutual suppression effects in SFOAE responses. Based on the use of transient-gated stimuli, time-frequency representations of the OAE responses enable measurement of cochlear delays and detection of possible multiple internal reflections. Re-

sults will be described in terms of the reflection emission theory of OAEs [Zweig and Shera, J. Acoust. Soc. Am. **98**, 2018–2047 (1995)] extrapolated to moderate stimulus levels, in which SFOAE nonlinearities are directly related to basilar-membrane nonlinearities. [Work supported by NIH (R01 DC003784, T32 DC00013).]

**2pPP5. Transient otoacoustic emissions and frequency selectivity.** Julie Hazelbaker and Lawrence Feth (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH 43210)

The relationship between transient evoked otoacoustic emissions (TEOAE) and frequency selectivity in the auditory system was examined. Three subjects with normal hearing sensitivity and TEOAE spectra with prominent peaks and valleys participated. Psychophysical tuning curves were measured at two pairs of frequencies: one where a robust emission occurs, and the other for an emission with a signal-to-noise ratio at least 6 dB SPL lower than the first. A forward masking paradigm with a 2AFC, 3 up – 1 down adaptive procedure was used to construct three point tuning curves. Quiet threshold for a given probe frequency was established first. The tip of the tuning curve was then obtained by determining the masker level for  $P(C)=79.1$  with a probe at 10 dB SL. Finally, the Q<sub>10</sub> frequencies were acquired by adapting the frequency of the masker, which was set 10 dB above the tuning curve tip [Spetner and Olsho, 1990]. Preliminary results indicate that the Q<sub>10</sub> values for the tuning curves from the TEOAE peak frequencies are consistently greater than those of the tuning curves from valley frequencies. These results suggest that a relationship exists between frequency selectivity in the auditory system and the relative minima and maxima of TEOAE patterns. [Research supported by a grant from The Ohio State University College of Social and Behavioral Science.]

**2pPP6. Wideband reflectance measurement of the contralateral acoustic reflex threshold for broadband noise and 1000-Hz tones.** M. Patrick Feeney and Lindsay Marryott (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210)

Recent work has suggested that wideband reflectance measures may provide a sensitive estimate of the acoustic reflex threshold [M. P. Feeney and D. H. Keefe, J. Speech, Lang. Hear. Res. **42**, 1029–1041 (1999)]. This study examined contralateral reflex thresholds for broadband noise and 1000-Hz tones in 21 young adults. The acoustic reflex was detected with the reflectance system by assessing the correlation between pairs of wideband reflectance shifts for a given activator level. Reflex threshold estimates obtained with this method were compared with those obtained with a clinical admittance system. Reflex thresholds for the noise activator were around 10 dB lower for reflectance ( $M=77.4$ -dB SPL, Zwislocki coupler) than for the clinical measure [ $M=87.8$ -dB SPL,  $t(20)=6.2$ ,  $p<0.001$ ]. For the 1000-Hz activator, reflectance reflex thresholds averaged 5 dB lower ( $M=95.3$ -dB SPL) than for the clinical system [ $M=100.2$ -dB SPL,  $t(20)=8.2$ ,  $p<0.001$ ]. Moreover, the tone-noise reflex threshold difference was significantly larger for reflectance ( $M=17.9$ -dB SPL) than for the clinical measure [ $M=12.4$ -dB SPL,  $t(20)=6.2$ ,  $p<0.001$ ]. These results support the use of wideband reflectance as a sensitive measure of the acoustic reflex threshold. [Supported by NIH-R03DC04129.]

**2pPP7. An auditory-nerve model with an instantaneous frequency glide and its application to modeling psychophysical results.** Qing Tan and Laurel Carney (Dept. of Biomed. Eng., Hearing Res. Ctr., Boston Univ., Boston, MA 02215)

A computational model was developed to simulate the responses of auditory nerve (AN) fibers. The model consists of a time-varying band-pass filter with a nonlinear feed-forward control path, which changes the bandwidth and gain of the signal path. This model produces realistic response features to various stimuli, including pure tones, two-tone combinations, wide-band noise and clicks. The parameters of the band-pass filter

were estimated by fitting the model revcor functions to revcor functions from experimental data of cat. The Marquardt method was used to minimize the difference between the model revcor function and the cat revcor function at various characteristic frequencies. Instantaneous frequency (IF) glides in the reverse correlation function of the model's response to broadband noise were achieved by simple configuration of the locations of the poles and zeros in the band-pass filter. The locations of the poles are continuously varied by the control signal to change the gain and bandwidth of the signal path, without affecting the IF profile, which is level independent. Other important properties, such as nonlinear compression, two-tone suppression and reasonable Q<sub>10</sub> values are also included. Applications of this model for studying AN responses to speechlike signals will be discussed.

**2pPP8. Encoding of sinusoidal and natural speech tokens in the auditory nerve.** Jeremy Loebach and Robert Wickesberg (Dept. of Psych., Univ. of Illinois, 603 E. Daniel St., Champaign, IL 61820)

Sinewave speech is a synthetic variation of natural speech that replaces natural formants with time varying sinusoidal waves. This synthesis preserves the general formant motion of the natural utterance while removing many spectral details; yet sinewave speech tokens can convey a linguistic message [Remez *et al.*, Science **212**, 947–950 (1981)]. This study examined the extent that phonetic information in sinusoidal speech is preserved at the level of the auditory periphery. We compared the responses of individual auditory nerve fibers in ketamine anesthetized chinchillas to the natural and sinusoidal tokens of the word /bal/. Comparisons of the first 100 milliseconds of the responses revealed high correlations over the first 20 milliseconds of the responses representing the initial consonant ( $r=0.82$ ) despite low correlations between the stimuli themselves ( $r=0.18$ ). Correlation coefficients decreased in the vowel portion of the responses ( $r=0.30$ ). Spectral reduction in the sinusoidal token may produce the low correlation between the neural representations of the vowel. The preservation of temporal cues may explain the high correlation between the neural representations of the consonant despite spectral differences.

**2pPP9. Consistency of the auditory nerve response to normal and whispered speech.** Stephanie Justin and Robert Wickesberg (Dept. of Psych., Univ. of Illinois, 603 E. Daniel St., Champaign, IL 61820)

Multiple presentations of an acoustic stimulus are often used to study the encoding of sounds that in the natural environment are not repeated. This study examined how consistent the response of the auditory nerve is with respect to the number of repetitions. Naturally voiced and whispered forms of /ta/ were presented 50 times at 60 and 90 dB to ketamine anesthetized chinchillas. For each stimulus, global average peri-stimulus time (GAPST) histograms for the first 20 trials were compared to those for the last 20 trials. Correlation coefficients for the normal syllable were 0.87 and 0.72 during the consonant and vowel, respectively. In the whispered condition, correlation coefficients were 0.95 during the consonant, but only 0.44 during the vowel. Similar correlations were obtained with comparisons of GAPSTs for either the first or last 20 trials to those for all 50 trials. Correlations obtained from the GAPSTs were higher than those of individual responses for both speech conditions and at all intensities. Individual auditory nerve fiber correlations during the speech stimuli were variable. These results demonstrate that the use of many repetitions to achieve consistency in the response may not be necessary when one studies the ensemble response of the auditory nerve.



**2pPP10. Stimulation rate influences frequency tuning characteristics of inferior colliculus neurons.** Jeremy Smalling, Alexander Galazyuk, and Albert Feng (Univ. of Illinois, Urbana, IL 61801)

Sounds in real world environments such as animal calls and human speech are complex and often occur in rapid succession. In light of this, it is important to gain an understanding of how the rate of acoustic stimulation influences the units' basic response properties. Previous studies of frequency tuning were generally based on investigating neuronal responses to tonal stimuli presented in isolation, and the results derived therefrom were assumed to accurately portray the cell's response range. However, the firing history of an auditory neuron has been shown to shape its response to subsequent sounds. The goal of the present study was to investigate the frequency tuning characteristics of neurons in the inferior colliculus (IC) of the little brown bat to tone pulses presented at various rates. Eighty-five percent of the IC neurons studied showed rate-dependent changes in their frequency selectivity. Half of these neurons exhibited narrowing of their frequency response range at higher rates. These results indicate that the response properties of central auditory neurons at low stimulation rates do not necessarily reflect the units' response properties to sounds presented at high and behaviorally relevant rates. The possible cellular mechanisms and behavioral importance of this effect are discussed.

**2pPP11. Noninvasive recording of fetal auditory evoked responses using a new 151 channel sensor array (SARA).** Hari Eswaran, James Wilson (Dept. of Appl. Sci., Univ. of Arkansas at Little Rock, Little Rock, AR 72204), Hubert Preissl, Pamela Murphy, and Curtis Lowery (Univ. of Arkansas for Medical Sci., Little Rock, AR 72205)

The goal was to successfully record fetal auditory evoked responses (AERs) using the newly developed noninvasive fetal magnetoencephalographic (MEG) system called SARA. 131 AER measurements were performed on 31 fetuses starting at 27 weeks. The SARA system, equipped with an array of 151 SQUID (superconducting quantum interference device) sensors spread over an area of 1300 cm<sup>2</sup>, is curved to fit the shape of the pregnant abdomen. The auditory stimulus was—1 kHz tone; duration—100 ms; rate—0.5 tones/s.; intensity (external)—100-dB SPL in air. The data were collected at a sampling rate of 312.5 Hz. The AERs were averaged over a period of 700-ms window after removing interfering maternal and fetal heart signals. Control recordings were performed on the same fetus with the same parameters but without any sound stimulation. The peak of the evoked responses was around 250 ms with amplitudes of about 20 fT. No evident evoked response was seen in the control recordings on the same fetus. In summary, it is feasible to noninvasively acquire MEG auditory evoked responses from the fetus using SARA. AER can be a useful tool for the neurological assessment of fetal well-being throughout gestation. [Work funded by NIH.]

**2pPP12. A coincidence-detection model for level discrimination of tones in roving-level noise.** Xuedong Zhang and Laurel H. Carney (Hearing Res. Ctr. and Dept. of Biomed. Eng., Boston Univ., Boston, MA 02215)

This study describes a monaural cross-frequency coincidence-detection mechanism to model level discrimination performance for a tone in the presence of noise in a roving-level paradigm. Model coincidence detectors receive population input from model auditory-nerve (AN) fibers from the same ear with different characteristic frequencies (CFs). The response of each model coincidence detector is sensitive to both the rate and the phase of its AN fiber inputs. A set of model coincidence detectors was constructed that received input from a population of model AN fibers. To detect intensity increments of tones in wideband noise, the most sensitive coincidence detector was one that received the inputs from two AN fibers with tone responses that were opposite in phase (phase opponency); this mechanism was robust in the presence of roving-level maskers. For narrow-band noise, detectors that were sensitive to changes in rate also contributed information to the detection task; this rate-based mechanism

was affected by a roving-level paradigm. The difference in the effect of the rove on model performance with different masker bandwidths was similar to that observed for human subjects [Kidd *et al.*, J. Acoust. Soc. Am. **86**, 1310 (1989)].

**2pPP13. A low-frequency paradox in binaural pitch.** William M. Hartmann (Phys. and Astron., Michigan State Univ., East Lansing, MI 48824) and Peter Xinya Zhang (Michigan State Univ., East Lansing, MI 48824)

Binaural pitches, such as the Huggins pitch and binaural coherence edge pitch (BICEP), can be approached in terms of a model network of binaural neurons that are tuned in frequency and in interaural delay [e.g., Raatgever and Bilsen, J. Acoust. Soc. Am. **80**, 429–441 (1986)]. The basic rule of binaural combination resembles addition. Binaural pitches appear as peaks or edge features in the frequency-delay plane. The characteristic frequency of a feature determines the pitch, and the internal delay of the feature determines the lateralization of the pitch image. Because delay lines become sparser with increasing delay, the model predicts that binaural pitches of very low frequency should be more difficult to hear the more they are lateralized. However, our Huggins pitch and BICEP discrimination and detection experiments show that the reverse is true. These experiments favor a model in which the basic rule of binaural combination resembles subtraction. However, the predictions of the subtractive model disagree with the observed lateralization. Furthermore, the low-frequency discrimination and detection experiments do not agree with a model in which additive and subtractive binaural combinations are alternative options. This is the low-frequency paradox. [Work supported by NIDCD.]

**2pPP14. The binaural processing of double vowels: Experimental measurements and computational models.** Michael A. Akeroyd (Dept. of Neurosci., Univ. of Connecticut Health Ctr., Farmington, CT 06030 and Lab. of Exp. Psych., Univ. of Sussex, Brighton, UK)

The degree to which the binaural processing of a "target" vowel is affected by the simultaneous presentation of a "distractor" vowel (of different ITD to the target) was studied using three pairs of synthetic vowels. The stimuli were an "er" (of 100-Hz *F*<sub>0</sub>) paired together with either an "ai," "er," or "oo" (all of 125-Hz *F*<sub>0</sub>). In Experiment 1 ITD thresholds for the target vowel were measured. The results showed that thresholds were larger in double-vowel pairs than in control, target-alone situations. Moreover, the size of the effect was dependent both on which vowel of the pair was the target and on the relative level of the two vowels. Computational modeling of the binaural correlograms of the stimuli showed that these results do not require any explicit segregation of frequency channels for the two vowels. In Experiment 2 the perceived lateralization of the target vowel (when given an ITD of 0 ms) was measured. The results showed that this lateralization was "pulled" slightly toward the distractor vowel. Further computational modeling showed that some segregation of the frequency channels in the binaural correlogram, perhaps based on the pitch information in the corresponding autocorrelogram, is required to account for this result.

**2pPP15. MLDs for signals placed in masker envelope minima and maxima.** Emily Buss, Joseph HallIII, and John Grose (Dept. of Otolaryngol./Head and Neck Surgery, Univ. of North Carolina School of Medicine, 610 Burnett-Womack Bldg., CB# 7070, Chapel Hill, NC 27599)

Previous data on the masking level difference (MLD) have suggested that NoSpi detection for a long-duration signal is dominated by signal energy occurring in masker envelope minima [Hall *et al.*, J. Acoust. Soc. Am. **103**, 2573–2577 (1998); Grose and Hall, *ibid.* **103**, 2590–2594 (1998)]. This finding was expanded upon using a brief (30-ms) 500-Hz tonal signal that coincided with either the envelope maximum or envelope minimum of a 50-Hz wide bandpass Gaussian noise masker centered at



500 Hz. On each interval a masker band was digitally generated in the frequency domain, the envelope maximum (or minimum) was identified, and the sample was rotated until that maximum (or minimum) fell in the temporal center of the sample. Data were collected at a range of masker levels (25, 40, and 55 dB spectrum level) in order to assess the possible contributions of compression. Conditions employing frozen noise samples were included to assess the possible role of masker uncertainty. Results to date suggest substantially larger MLDs for signals coinciding with masker envelope minima as contrasted with envelope maxima. This result is primarily due to decreased NoSpi thresholds for signals coinciding with minima. [Work supported by NIH NIDCD.]

**2pPP16. Effect of reverberation on the masking of narrow-band signals.** Richard Freyman, Uma Balakrishnan (Dept. of Commun. Disord., Univ. of Massachusetts, Rm. 3 Arnold House, Amherst, MA 01003), and Patrick Zurek (Sensimetrics Corp., Somerville, MA 02144)

This study explores the factors that influence the masking of narrow-band signals in reverberant sound fields. Rectangular rooms were simulated using the image method with a rigid sphere modeling the head. The signals were third-octave noise bands presented in a continuous broadband noise masker. For a given room simulation, the signals and masker were convolved with impulse responses obtained in the room at multiple azimuth angles and distances, for subsequent presentation via headphones. Adaptive forced-choice procedures were used with the resulting signals to find monaural and binaural thresholds in listeners with normal hearing. The benefit of separating the masker and signal spatially in the simulation was reduced by reverberation, as expected, although the effects were complex. Among the predictions confirmed by these experiments is that reverberation actually improves thresholds when the masker is closer to the listener than the signal, apparently due to both an increase in monaural signal power a decrease in interaural signal correlation. With masked thresholds obtained over the frequency range of the articulation index, the results of these experiments will allow quantitative predictions of speech intelligibility as a function of room characteristics and speech and noise source locations. [Work supported by NIH.]

**2pPP17. Short-term adaptation to novel combinations of acoustic spatial cues.** Timothy Streeter, Barbara G. Shinn-Cunningham, and Andrew Brughera (Hearing Res. Ctr., Dept of Cognit. and Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215, timstr@cns.bu.edu)

Previous work demonstrates that a listener's interpretation of spatial cues can change due to long- or short-term training. Results suggest that short-term training does not alter how spatial locations are computed, but how spatial percepts are interpreted. This series of experiments examines how subjects adapt to different rearrangements of auditory cues. In Experiment I, subjects were trained to identify the spatial cues normally associated with exocentric location  $x$  as coming from  $2x$  (a pure linear remapping of space). In Experiment II, ITDs were remapped in the same way, but were paired with normal IID and spectral cues (i.e., the ITD normally associated with azimuth  $2x$  was paired with IID and spectral cues associated with location  $x$ ). Results examine how spatial resolution and bias are affected by unusual mappings between spatial cues and exocentric location as well as by combining spatial cues that normally do not correspond to one source location. These experiments test the hypothesis that listeners rapidly and subconsciously recalibrate a "gain" describing how to map an internal representation of auditory space to exocentric space, but cannot rapidly alter how the source positions in this internal representation are computed. [Work supported by a grant from the Whitaker Foundation.]

**2pPP18. The independence of adaptation to interaural time and level changes.** Michael F. Neelon, Carrie M. Benedon, and Rick L. Jenison (Dept. of Psych., Univ. of Wisconsin, Madison, WI 53706)

Changes in interaural time and interlevel differences (ITDs and ILDs) for a sound moving in the free field normally imply movement in the same direction. Research has shown that listeners can be adapted to and show aftereffects for each of these cues separately [W. H. Ehrenstein, *Perception* **23**, 1249–1255 (1994)]. Psychophysical evidence of ILD to ITD conversion, however, raises the question whether adaptation is independent between these two channels. The present experiment explores their possible independence in order to better specify where in the auditory pathway their adaptation occurs. Listeners will be adapted to sounds with dynamically varying ITD and ILD cues alone, in concert or in conflict with each other (the latter conditions implying either the same or opposite direction of movement). ITD and ILD adaptation will be measured separately via test sounds changing in either of these cues alone. If aftereffects for one cue are unaffected by the implied movement direction of the other, then adaptation most likely lies early in the auditory pathway (e.g., the olivary bodies). However, if aftereffects show an influence of one cue on the other, some later physiological site of convergence may be involved in adaptation (e.g., the inferior colliculus).

**2pPP19. The role of the precedence effect in sound source lateralization.** Daniel E. Shub (Speech and Hearing Sci. Prog., Harvard–MIT Div. of Health, Sci. and Technol.), Robert H. Gilkey (Wright State Univ.), and H. Steven Colburn (Boston Univ.)

This work examines sound source lateralization of ongoing broadband noise targets in the presence of a single ongoing broadband noise jammer, which was either a simple simulated reflection of the target or was uncorrelated with the target. Identification thresholds of  $\pm 300\text{-}\mu\text{s}$  ITD targets and lateralization thresholds were determined. Identification thresholds of the reflection jammers were 3.2 dB lower ( $P = 0.027$ ) than the identification thresholds of the jammers that were uncorrelated with the targets. Lateralization thresholds of the reflection jammers were 2.85 dB lower ( $P = 0.040$ ) than the lateralization thresholds of the uncorrelated jammers. There was a slight trend, in agreement with the current understanding of the precedence effect, for both the identification and lateralization thresholds of the simple reflection jammers to be dependent on jammer ITD. The  $0\text{-}\mu\text{s}$  ITD reflection jammer produced a threshold that was 1.9 dB lower ( $P = 0.104$ ) than the threshold of the  $643\text{-}\mu\text{s}$  ITD reflection jammer. Additionally comparisons between the lateralization thresholds of normal-hearing listeners and two cross-correlation models were made. The models obtained lateralization thresholds as low as  $-13$  dB, up to 10 dB better than normal-hearing performance. [Work supported by NIH Grant No. R01 DC00100.]

**2pPP20. The minimum audible facing angle.** John G. Neuhoff, Mary-Alice Rodstrom, and Tanaya Vaidya (Dept. of Psych., The College of Wooster, Wooster, OH 44691, jneuhoff@wooster.edu)

Many sound sources have a directional characteristic such that they project sound from one planar orientation at a time (e.g., talkers and loudspeakers). Yet, almost all localization studies employ a unidirectionally facing source aimed at the listener. Although there are good reasons for holding the facing direction of a loudspeaker constant relative to the listener when investigating localization, it is curious that the acoustic facing direction itself has received little or no attention. Here, blindfolded listeners heard pulsed noise from a loudspeaker facing a reference point of either 0 deg (directly facing the listener) or 30 deg. The loudspeaker was then rotated clockwise or counterclockwise to one of six facing angles (5, 10, 15, 20, 25, or 30 deg), and the stimulus was replayed. The listeners indicated perceived direction of rotation. The minimum threshold for detection, or in this case what is termed the minimum audible facing angle (mAFA), was 15 deg for the 0 deg reference point and 21 deg for the 30

deg reference point. Within the limitations of the current experimental conditions, listeners showed appreciable sensitivity to the facing angle of a unidirectionally facing sound source. Our results also show evidence for a maximum auditory facing angle (MAFA).

#### **2pPP21. Effect of auditory cuing on azimuthal localization accuracy.**

Norbert Kopčo (Hearing Res. Ctr., Dept. of Cognit. and Neural Systems, Boston Univ., 677 Beacon St., Boston, MA 02215, kopco@cns.bu.edu), Albert Ler, and Barbara G. Shinn-Cunningham (Boston Univ., Boston, MA 02215)

Auditory localization in the horizontal plane was measured following the presentation of a cue in order to explore whether attentional focus could improve localization accuracy. Subjects pointed to the heard location of a broadband target source that was presented (at a delay of either 50 or 300 ms) after a cue source. In half of the blocks of trials, the cue source came from the same (left/right) hemifield as the target on most (75%) of the trials, and thus (on average) provided the listener with information about the target location. In the other half of the blocks of trials, the cue source location was equally likely to come from either the same or the opposite hemifield and provided no information to the subject regarding target position. The presence of a cue biased localization performance in both conditions rather than improving accuracy when the cue provided information about the target laterality, even for a delay of 300 ms between the cue and target. Results suggest that auditory cuing, which has been shown to decrease response times, degrades localization accuracy. [Work supported by a grant from the Air Force Office of Scientific Research.]

#### **2pPP22. Localization and speech-identification ability of hearing-impaired listeners using phase-preserving amplification.**

Ward Drennan, Stuart Gatehouse, Patrick Howell (MRC Inst. of Hearing Res., Glasgow Royal Infirmary, Queen Elizabeth Bldg., Glasgow G31 2ER, UK), Dianne VanTasell, and Steven Lund (Starkey Labs., Eden Prairie, MN 55344)

Hearing-impaired listeners experience increased difficulties recognizing speech and localizing sounds in adverse environments. This study investigated the benefits of signal processing in binaural hearing aids designed to preserve cues that accompany spatial location. The ability of listeners to localize click-trains in noise was tested, along with their ability *both* to localize *and* to identify words in noise (a dual task). Listeners experienced two types of bilateral hearing aid fittings: (1) a custom fitting that provided appropriate gain while also matching the phase measured near the tympanic membranes without the hearing aids, and (2) a conventional fitting (using the same hearing aid device) that provided the same gain with noncustom, linear phase. Testing occurred for each fitting immediately and following 3-months listening experience using a within-listener, within-device, randomized, single blind, crossover design. A rating-scale questionnaire was administered to assess perceived speech-hearing and spatial abilities. In the dual task, the listeners exhibited superior localization ability for the phase-preserving fitting initially and after 3-months experience. This advantage did not occur for the click-train localization task. Listeners rated their spatial abilities higher with the phase-preserving fitting, although little improvement was observed or reported for speech hearing.

#### **2pPP23. Localization of sound by binaural cochlear implant users.**

John P. Preece, Richard S. Tyler, Jay T. Rubinstein, Bruce J. Gantz (Dept. of Otolaryngol., Univ. of Iowa, Iowa City, IA 52242), and Richard J. M. van Hoesel (CRC for Cochlear Implant and Hearing Aid Innovation, E. Melbourne 3002, Australia)

We examined the localization ability in five adult patients who were implanted bilaterally with the CI24M implant from Cochlear Corporation. These patients demonstrated a difference in either length of time deaf before implantation, preimplant thresholds, or both. Patients were tested in

an anechoic room. Signals were four 200-ms bursts of broadband noise separated by 55 ms of silence. Stimuli were randomly presented from one of eight loudspeakers arrayed in an arc at ear level in front of the patient. The speakers were separated by 15 deg azimuth. The patient was seated 1.5 m from the speakers and responded orally with a speaker number. The level of individual stimuli was varied randomly over an 8-dB range with an average level of presentation of 65-dB SPL measured at the approximate location of the center of the patient's head. Patients were tested with each ear separately and with both ears together. The results show a very good ability in all five patients to localize sounds with two cochlear implants. The monaural abilities varied considerably across patients, and often between ears for each patient, but were always worse than the binaural abilities. [Work supported by NIDC and CRC.]

#### **2pPP24. Auditory motion aftereffects with varying interaural phase difference.**

Takayuki Kawashima and Takao Sato (Dept. of Psych., Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-0033, Japan, ll97031@mail.ecc.u-tokyo.ac.jp)

It has been known that the auditory motion aftereffect (MAE) is spatially specific [see, for example, D. W. Grantham, *Acoustica* **84**, 337–347 (1998), exp. 3]. However, it still is not very clear which cues for sound localization are responsible for this spatial specificity of the MAE, since several possible cues, such as the spectral profile or interaural level difference, covaried in most past studies. In this study, MAE and its spatial specificity were investigated with sound motion created by using only interaural phase difference (IPD) as a first step to identify the responsible cues. We used the probe method with the method of constant stimuli. Either with or without adaptation to a moving tone, subjects were asked to judge the direction of probe tone motion (0.7 ms duration, either to the left or the right). The slope and the position of the psychometric functions were affected by the direction of the adapter's motion direction, but only when IPD ranges (spatial ranges of motion) of the adapter and probe tones overlapped each other. These results suggest that the change in IPD is at least one of the cues which produce the spatial specificity.

#### **2pPP25. Minimum dynamic lateralization for multiple moving sources.**

Michael F. Neelon and Rick L. Jenison (Dept. of Psych., Univ. of Wisconsin, Madison, WI 53706)

It is still unresolved whether auditory motion is perceived via specialized motion detectors or inferred from static samples of changes in spatial position. This question has been investigated by comparing minimum audible movement angles for single sources moving discretely or continuously [Perrott and Marlborough, *J. Acoust. Soc. Am.* **85**, 1773–1775 (1989)]. But using one source does not force the listener to determine movement by only one of the aforementioned processes. To better measure sensitivity to pure auditory dynamics, the following study simulates multiple overlapping sources moving in the same direction across random sections of the auditory hemifield. Experimental stimuli are composed of different portions of the trajectories of circling resonant sources, which are individually created by dynamically varying interaural time and level differences. The multiple, variable endpoints in the composite stimulus should inhibit the listener from relying solely on such cues to determine movement direction. Pilot studies using from 1–5 concurrent sources show dynamic lateralization thresholds are lowest for one moving source. This implies movement is best perceived when stimulus endpoints can be sampled. However, thresholds across multiple sources do not significantly differ regardless of number, which may represent a measure of sensitivity to pure auditory dynamics.

**2pPP26. Finite-difference computation of head-related transfer functions for human hearing.** Tian Xiao and Qinghuo Liu (Dept. of Elec. Eng., Duke Univ., Durham, NC 27708-0291)

Modeling the head-related transfer function (HRTF) is a key to many applications in spatial audio. To understand and predict the effects of head geometry and the surrounding environment on the HRTF, a three-dimensional finite-difference model has been developed for the head-related impulse response. A perfectly matched layer (PML) is used to

absorb outgoing waves at the truncated boundary of an unbounded medium. An external source is utilized to decrease the computational domain size. This numerical model has been validated by analytical solutions for a spherical head model. The FD code is then used as a computational tool to predict the HRTF for various scenarios. In particular, a simplified spherical head model is compared to a realistic head model up to about 7 kHz. The HRTF is also computed for a realistic head model in the presence of a wall. It is demonstrated that this FD model can be a useful tool for spatial audio applications.

TUESDAY AFTERNOON, 5 JUNE 2001

SALON V, 1:15 TO 5:00 P.M.

## Session 2pSA

### Structural Acoustics and Vibration: Scattering, Plates and Shells

Sean F. Wu, Chair

*Department of Mechanical Engineering, Wayne State University, 505 Anthony Wayne Drive, Detroit, Michigan 48202*

#### Contributed Papers

1:15

**2pSA1. Interaction of Lamb wave with a welded rib.** Bruno Morvan, Jean Duclos, Hugues Duflo, and Alain Tinel (Laboratoire d'Acoustique Ultrasonore et d'Electronique (LAUE), UMR 6068 CNRS, Pl. R. Schuman, BP 4006, 76 610 Le Havre, France)

This work deals with Lamb wave conversion on a rib stiffened plate. Some previous work has studied the scattering of plane waves at a  $T$  junction at low frequency ( $FE < 200$  kHz mm) for which the wavelengths are higher than the plate thickness. In this study, the conversion phenomenon is investigated in the case of an A1 incident Lamb wave ( $FE = 4$  MHz mm). The experimentation is done with a stainless steel  $T$  structure, and the plate and rib are 2 mm thick. The normal surface displacement is related theoretically to the power flow. Then, the energy of the converted waves is obtained from measurement of the plate normal displacements by means of a laser interferometer. Next, we compute a finite-element simulation on two various meshes: the plates are connected perpendicular to each other either with or without a weld. The comparison between experimental and numerical methods shows the importance of the weld shape on the wave transmission to the rib. The weld funnels the incident wave and improves the energy transmission on the rib. Moreover, the experimental and MEF results are in a good agreement.

1:30

**2pSA2. Homogenization of a ribbed cylindrical shell over the full frequency range for efficient calculation of structural-acoustic behavior.** Linda Franzoni and Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708-0300)

A homogenization method for complex structures, applicable for all frequencies, is described for a cylindrical shell with periodically spaced ribs. This approach has application to naval and aerospace structures. The homogenization method utilizes a local-global decomposition facilitated by adding and subtracting canceling smooth forces. The smooth global problem has an infinite-order structural operator, and periodic discontinuities are replaced by equivalent distributed suspension terms. The global problem can be solved very efficiently since all rapidly varying scales have been removed. The local problem, which provides transfer function information for the global problem, is solved separately and independently, except for amplitude information from the global problem. Once formulated, the self-contained global problem is solved first, and the local solution can be reconstructed afterwards. In the present work, the ribs are modeled as annular plates periodically attached to the cylindrical shell. The coupled shell equations are re-cast using the method of local-global

homogenization, and are solved efficiently using a combination of symbolic manipulation and numerical methods. Comparisons with other solution methods are presented, and the extension to fluid loading is discussed. [Work sponsored by ONR.]

1:45

**2pSA3. Improvement of the solution convergence rate for an elastic shell driven by a vibratory constraint using analytical-numerical matching.** Christopher Park, Linda Franzoni, and Donald Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708-0300)

A three-dimensional elastic shell is driven by the motion of an attached support and the solution is constructed using the method of analytical-numerical matching (ANM) with finite-element analysis and the structural-acoustic scattering code SONAX. First, the area around the structural constraint is resolved in great detail as a separate local three-dimensional FEA solid model. Second, a Love-Timoshenko shell-type model of the local region is solved analytically for a set of smooth forces that will be applied to the full three-dimensional shell in place of the constraint. Third, the model of a full three-dimensional shell with end caps is analyzed in SONAX, with the smooth forces applied on a mode-by-mode basis. The complete solution to the original problem is the superposition of the solutions of each of the subproblems described. Note that the high modal content of the original problem is contained in the high-resolution separate local problem (small domain), leaving only low modal content in the global (large domain) problem that is solved using SONAX. This sample problem demonstrates the ability of ANM to enhance the capabilities of SONAX and to improve the convergence rate (and overall computation time) for problems of this type. [Work sponsored by ONR.]

2:00

**2pSA4. Circumferential-wave phase velocities for empty, fluid-immersed spherical metal shells.** Herbert Überall (Phys. Dept., Catholic Univ. of America, Washington, DC 20064), A. Claude Ahji, P. K. Raju (Auburn Univ., Auburn, AL 36849-5341), I. K. Bjørnø, and L. Bjørnø (Tech. Univ. of Denmark, DK-2800 Lyngby, Denmark)

Our earlier studies regarding acoustic scattering resonances and the dispersive phase velocities of the surface waves that generate them, have demonstrated the effectiveness of obtaining phase velocity dispersion curves from the known acoustic resonance frequencies, and their accuracy. This possibility is offered by the picture of phase matching after each complete circumnavigation of these waves, which leads to close agreement of resonance results with those calculated from 3-D-elasticity theory



whenever the latter are available. The present investigation is based on the mentioned resonance frequency/elasticity-theory connection, and we obtain comparative dispersion-curve results for water-loaded, evacuated spherical shells of various metals. In particular, the characteristic upturn of the dispersion curves of low-order shell-borne circumferential waves ( $A$  or  $A_0$  waves) which takes place on spherical shells when the frequency tends towards low values, is demonstrated here for all the metals under consideration.

## 2:15–2:30 Break

### 2:30

**2pSA5. Visualization of the energy flow for a guided forward wave in and around a fluid loaded elastic cylindrical shell.** Cleon E. Dean (Phys. Dept., P.O. Box 8031, Georgia Southern Univ., Statesboro, GA 30460-8031, cdean@gasou.edu) and James P. Braselton (Georgia Southern Univ., P.O. Box 8093, Statesboro, GA 30460-8093)

Previous work [Cleon E. Dean and James P. Braselton, *J. Acoust. Soc. Am.* **107**, 2921 (2000)] showed the energy flow for a fluid loaded elastic cylindrical shell at the resonance frequency. The results were equivocal since there are two counterpropagating guided waves. The current work uses a clever analysis due to Kaduchak and Marston [Gregory Kaduchak and Philip L. Marston, *J. Acoust. Soc. Am.* **98**, 3501–3507 (2000)] to separate the two waves and show the energy flux associated with just one forward propagating surface guided wave.

### 2:45

**2pSA6. Scattering and radiation from a fluid-loaded coated cylindrical shell.** J. M. Cuschieri and C. Treffot (Ctr. for Acoust. and Vib., Dept. of Ocean Eng., Florida Atlantic Univ., Boca Raton, FL 33431)

A common question when dealing with fluid-loaded coated structures is how well a normally reacting coating layer models the radiation and scattering from a fluid-loaded coated cylindrical. Modeling with a normally reacting layer simplifies the model which therefore allows more complex conditions to be analyzed. To explore this question analytic solutions are obtained for the scattering from a fluid-loaded cylindrical shell with a compliant coating using both multilayer shell theory [L. Flax and W. G. Neubauer, *J. Acoust. Soc. Am.* **61**, 307 (1977)], and a normally reacting coating solution [J. M. Cuschieri and D. Feit, *J. Acoust. Soc. Am.* **107**, 3196 (2000)]. The similarities and differences in the scattering results that are obtained using either of these two approaches are presented for different coating thicknesses and characteristics.

### 3:00

**2pSA7. Cepstrum of the scattering response of a cylinder.** Angie Sarkissian and Louis R. Dragonette (Naval Res. Lab., Washington, DC 20375-5350, angie@aquanrl.navy.mil)

The cepstrum of the scattered field of a ribbed, hemispherically end-capped cylindrical shell is computed for various incident and scattered angles, for  $2 \leq ka \leq 30$ , where  $k$  is the wave number and  $a$  is the radius of the cylinder. Highlights observed in the cepstrum domain are related to various scattering mechanisms. The cepstrum representation is compared to the frequency domain and the time domain response of the cylinder. Benefits and disadvantages of the cepstrum representation are discussed. [Work supported by ONR.]

### 3:15

**2pSA8. Acoustic scattering from a fluid-loaded membrane with periodic discontinuities using local-global homogenization (LGH).** Donald Bliss, Linda Franzoni, and Pavel Danilov (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708-0300)

A homogenization method for complex structures, applicable over the entire frequency range, is being developed. The method utilizes a local-global decomposition facilitated by adding and subtracting canceling

smooth forces. The low-wave-number global problem has an infinite-order structural operator, and periodic discontinuities are replaced by an equivalent distributed suspension. The rapidly varying local problem, which provides transfer function information for the global problem, is solved separately and independently, except for amplitude information from the global problem. Once formulated for a specific structure, the self-contained global problem is solved first, and the local solution can be reconstructed afterwards. The LGH reformulation allows the global problem to be solved at much lower resolution than the length of flexural waves on the original structure. The application of this approach to scattering from a fluid-loaded membrane is described. The effects of fluid radiation are transferred entirely to the smooth global problem, whereas evanescent fluid modes are contained within the local problems. Sample calculations are presented comparing the method with classical solutions. [Work supported by ONR.]

## 3:30–3:45 Break

### 3:45

**2pSA9. Large coated plates under random excitation: Spatial-domain insights into radiation efficiency.** R. Martinez and J. M. Garrellick (Cambridge Acoustics Assoc./Anteon Corp., 84 Sherman St., 3rd Level, Cambridge, MA 02140, rmartinez@caa.atinc.com)

A formal spatial-domain analysis of a large plate's cross-spectral response to TBL-like excitations yields the following two results, unified and in closed form: (1) a part that contains a moderate area effect (from the acoustic field's point of view, despite the plate being below coincidence by choice) and a stronger edge effect; and (2) a second part with a weak edge effect and a potentially strong area effect. The strong-edge/moderate-area part contains as coefficient the inverse of the plate's effective loss factor. The strong-area part does not, enabling it to dominate the former with rising frequency particularly over that range in the frequency spectrum encompassing the typically broad (lossy) thickness resonances of the plate's coating. Use of this unified cross-spectral response function to compute the plate's acoustic field confirms the expected rise in radiation efficiency at higher loss factors and frequencies, as brought about by the relative growth of the strong-area effect under both of those conditions. [Work supported by NSWC/Carderock Division.]

### 4:00

**2pSA10. Reconstruction of transient sound radiation from impulsively accelerated objects.** Manjit Singh and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

This paper presents reconstruction of transient acoustic radiation from impulsively accelerated objects using the HELS method [Wu, *J. Acoust. Soc. Am.* **107**, 2511–2522 (2000)]. The radiated acoustic pressure is expanded in terms of the spherical wave functions and spherical harmonics in the frequency domain. The coefficients associated with these expansion functions are determined by matching the assumed solution to the measured acoustic pressures. The errors incurred in this process are minimized by the least squares. Once the frequency-domain solution is obtained, the transient acoustic pressure signal is reconstructed by taking an inverse Fourier transformation via the residue theorem. The objects considered include an explosively expanding sphere, impulsively accelerated rigid sphere, and impulsively accelerated baffled sphere. The acoustic pressure signals thus obtained are compared with the analytic solutions. It is shown that this methodology can be extended to a nonspherical object subject to an arbitrarily time-dependent excitation. The resulting transient acoustic pressure can be reconstructed by a convolution integral of the impulse response function to the time history of the measured acoustic pressure signals. [Work supported by NSF.]

4:15

**2pSA11. The onset of the flexural mode in fluid loaded elastic shells revisited, and its value in object identification in both the frequency and time domain.** H. Uberall (Dept. of Phys., The Catholic Univ. of America, Washington, DC 20064) and M. F. Werby (Stennis Space Center, MS 39529)

The onset of the excitation of flexural resonances for fluid loaded evacuated elastic shells produces a striking event. This issue has been interpreted theoretically in 1988 by means of a partial wave decomposition which showed a very narrow-peaked subsonic wave and a broad-peaked weak wave that begins at the speed of sound of the entraining fluid and increases. The narrow peaks are identified with subsonic water-borne waves that resonate in the fluid along the surface of an elastic object, and the broad peaks correspond to the inception of flexural waves. They exist over a small interval in wave number at the point when the flexural wave begins to couple with the fluid. Here we examine the pulse solution. With increasing frequency the partial waves change phase (a 90 degree phase change) and the overlapping flexural waves transition from a partially coherent constructive signal to one that is partially destructive. This leads to an envelope or hump effect, also called mid-frequency enhancement; it is a function of shell thickness as well as material property. We demonstrate how this effect may be employed to identify a submerged elastic shell in either the pulse or frequency solution.

4:30

**2pSA12. The importance of the first and second symmetric Lamb modes in target identification of submerged elastic shells.** M. F. Werby (NRL Code 7180, Stennis Space Center, MS 39529 and Dept. of Phys., Catholic Univ. of America, Washington, DC 20064) and H. Uberall (Dept. of Phys., Catholic Univ. of America, Washington, DC 20064)

Resonances excited on any shell of constant thickness are one large resonance at a frequency inversely proportional to the shell thickness. Hence for thin shells it occurs at very high frequency. This effect occurs at the inception of the S1 resonance, which may be shown to be an

amplitude-modulated wave at inception. Its critical frequency may be determined by the condition:  $ka=3.14 \text{ VL}/(2 \text{ daVW})$  or  $ka=3.14 \text{ VTa}/(\text{daVW})$  where VW, VT, VL, a, da, and k are the speed of sound in the ambient fluid, transverse and longitudinal velocities in the elastic material, largest dimension of the object, object thickness relative to a and the wave number in the fluid. If  $2\text{VT}>\text{VL}$  the first condition defines the S1 critical frequency and the second that of the S2. The converse is true otherwise. The S2 resonance is not striking but may be identified as such. Thus, ratios of the two resonances lead to ratios of the bulk velocities and other considerations can isolate the shell thickness. This offers for any shell of constant thickness a means to determine the presence of certain submerged objects. We discuss the reasons for this and illustrate results in both time and frequency domains.

4:45

**2pSA13. Eigenfrequencies of fluid waves entrained in elastic shells.** H. Uberall (Dept. of Phys., The Catholic Univ. of America, Washington, DC 20064) and M. F. Werby (Stennis Space Center, MS 39529)

Acoustic signals that scatter from an elastic shell may excite elastic shell resonances. If the shell is evacuated then these as well as the pseudo-Stoney resonances are the only ones allowed. However, if there is a fluid inside then the eigenfrequencies of the enclosed fluid may be excited. Further, the presence of the fluid may alter the elastic body resonances. When the impedance of the enclosed fluid is not much smaller than the mechanical impedance of the elastic material then the scattered signal is greatly influenced by the enclosed fluid, but only at the allowed eigenfrequencies. Since the fluid is enclosed this leads to a discrete spectrum and an eigenvalue problem which is quite manageable. In this work we outline a method for determining the eigenfrequencies as well as their nature and use their values to isolate the actual body resonances due to the elastic material. We also illustrate the influence of the presence of the enclosed fluid on the elastic resonance frequencies.

TUESDAY AFTERNOON, 5 JUNE 2001

STATE BALLROOM, 1:00 TO 5:00 P.M.

## Session 2pSC

### Speech Communication: Dynamics of Speech Production and Perception

Pierre L. Divenyi, Chair

*Speech and Hearing Research, VA Medical Center, 150 Muir Road, Martinez, California 94553*

Chair's Introduction—1:00

#### Invited Papers

1:05

**2pSC1. Reconciling static and dynamic aspects of the speech process.** Björn Lindblom (Dept of Linguist., Univ. of Stockholm, S-10961 Stockholm, Sweden, lindblom@ling.su.se) and Randy L. Diehl (Univ. of Texas, Austin, TX 78712)

In speech, movement and spectral change are pervasive whereas steady-states are rare. Yet descriptive frameworks (e.g., IPA) focus mainly on static properties. Similarly, in quantitative models, dynamics is generally attributed to response characteristics while the underlying control consists of static units, e.g., spatial "targets," "equilibrium points," or "stable auditory goals." Accordingly, the input to the production of a diphthong would not be some dynamic aspect of the event, but an ordered string of two articulatory states. This approach creates a paradox when placed in the context of findings on visual and auditory perception that show that perceptual systems are more sensitive to changing stimulus arrays than to purely static ones. Clearly nervous systems have evolved to detect change. We are led to ask: If perception likes change, why assume production control in terms of static targets? We maintain that the two perspectives can be reconciled by according a significant role to positional targets in movement control and, at the same time, fully acknowledging the importance of dynamics in perception. In support of this position we present some recent computational work on simulating vowel systems by selecting optimal sets of vowels seen as objects of spectral change. [Work supported by NIH.]



**2pSC2. Dynamic units in speech production: Evidence from speech production errors.** Louis Goldstein (Dept. of Linguist., Yale Univ. and Haskins Labs., 270 Crown St., New Haven, CT 06511, louis.goldstein@yale.edu) and Dani Byrd (Univ. of Southern California, Los Angeles, CA 90089-1693)

Measurements of vocal tract activity during speech production show many distinct parts, all exhibiting different patterns of near-continuous motion. This description stands in sharp contrast to the description of speech as composed of a small number of discrete phonological units. In recent years the tools of dynamical systems have been brought to bear on the problem of relating these descriptions. The continuous vocal tract activity can be theoretically decomposed into discrete units of action, or gestures, each of which is a dynamical control regime for vocal tract articulators. These gestures can also serve as primitive phonological elements. Evidence for gestures as units of the speech production process has recently been obtained in experiments that measure the kinematics of speech errors produced when a simple phrase (e.g., “cop top”) is repeated. In errors, one of the gestures comprising the phrase may be duplicated in an anomalous position in the utterance. Since the anomalous gesture is often reduced in magnitude, such errors could not be detected without articulatory movement data. The nature of the errors and their asymmetries are consistent with the hypothesis that they represent phase transitions of a system of coupled dynamic units. [Work supported by NIH.]

2:05

**2pSC3. Production, coproduction and perception of speech gestures.** René Carré (Ecole Nationale Supérieure des Télécommunications, Département Signal, 46 rue Barrault, 75634 Paris cedex 13, France, carre@sig.enst.fr)

An acoustic model was used to derive distinctive speech gestures that result in dynamic deformations of the area function of an acoustic tube. These deformations are efficient, i.e., they provide maximum acoustic contrasts. It can be demonstrated that the gestures generated by the model are consistent with those in natural speech production. Using sequential and/or parallel combination of gestures,  $V_1V_2$  and  $V_1CV_2$  utterances were generated and presented to listeners to assess the effect of a particular coproduction scheme on phonemic identification. Perception of some gesture combinations remained remarkably invariant despite considerable variations of coproduction parameters: In a  $V_1CV_2$  token, for example, the onset of the consonant closure *re/* the  $V_1V_2$  transition can change to a large degree without altering the percept. In contrast, tokens such as [aya], obtained with two coproduced gestures (one tongue and one labial), require strict intergesture timing when [y] is just reached, in order for the French labial /y/ to be perceived. A comparison of the two classes of coproduction schemes and their perceptual consequences suggests possible ways of extracting, from formant trajectories (in the  $F1$ - $F2$  plane or the  $F1$ - $F2$ - $F3$  solid) represented in the time domain, acoustic cues that may underlie a number of coproduced gestures.

2:35–2:50 Break

2:50

**2pSC4. The synergy between speech production and perception.** Shihab A. Shamma (Dept. of Elec. Eng., Inst. for Systems Res., Univ. of Maryland, College Park, MD 20742, sas@eng.umd.edu)

Speech intelligibility is relatively unaffected by certain spectro-temporal deformations of the signal spectrogram. These include spectral and dynamic translations, stretching or contracting dilations, and shearing of the spectrogram along its temporal or logarithmic frequency axis. We shall argue that this stability results from a synergy between the dynamics and characteristics of the auditory cortex (the receiver) and of the vocal tract (the source). Specifically, on the perception side, neurophysiological evidence suggests that the acoustic spectrogram is represented in the primary auditory cortex along multiple spectro-temporal scales, and that the above deformations correspond to simple translations of the representation of the dynamic acoustic spectrum along these axes. On the production side, these spectro-temporal distortions can be caused by common variations among talkers in speaking rates, slurred articulatory dynamics, vocal tract length, cross-sectional profile, and losses. Using a simplified sinusoid model of the vocal tract, it is possible to relate “articulatory” parameters (e.g., the extent and location of a vocal tract constriction) directly to the formants, and hence to the cortical representation of the spectrum. A similar remarkable synergy exists between the dynamics of perception (e.g., via the perceived spectro-temporal modulation transfer functions) and the speed of articulation and hence the syllabic rates of speech. These results suggest an intimate link between perception and production, a view which offers new insights into the organization of speech phonemes (such as vowels) in terms of different vocal tract configurations.

3:20

**2pSC5. Vowel identity change: Trajectory length, transition velocity, or effort?** Pierre L. Divenyi (Speech and Hearing Res., V.A. Medical Ctr., Martinez, CA 94553, pdivenyi@ebire.org)

To what degree does the vocal tract shape need to be changed for a listener to perceive a vowel change? A 100-ms vowel  $V_1$  was followed by a transition toward vowel  $V_2$ .  $V_1$  and  $V_2$  were two vowels chosen from the triplet /a/, /i/, and /u/. Subjects had to indicate their confidence level that the transition, in fact, reached  $V_2$ . Data were converted to response-operating-characteristic (ROC) curves from which the transition velocity and transition duration were calculated for a fixed criterion of  $V_2$  percept. Results indicated that, regardless of the actual  $V_1V_2$  trajectory, the criterion was reached when the product of transition duration and transition velocity (=the trajectory length) was approximately constant. Since there is no logical explanation that could account for an invariance of trajectory length across all  $V_1V_2$  pairs, and since the criterion was reached with transition velocities that are not invariant, it appears that a constant percept is obtained when the transition encounters an elastic, opposing force that is inversely proportional to the acceleration at the start of the transition. This explanation, based on inferred articulatory effort, suggests the existence of a link between production and perception dynamics. [Work supported by NIH and the VA Medical Research.]

**2pSC6. What are the essential cues for understanding spoken language?** Steven Greenberg (Intl. Computer Sci. Inst., 1947 Center St., Berkeley, CA 94704, steveng@icsi.berkeley.edu)

Classical models of speech recognition (by both human and machine) assume that a detailed, short-term analysis of the acoustic signal is essential for accurately decoding spoken language. Several lines of evidence call this assumption into question: (1) intelligibility is relatively unimpaired when the frequency spectrum is distorted under a wide range of conditions (including cross-spectral asynchrony, reverberation, waveform time reversal and selective deletion of 80% of the spectrum), (2) the acoustic properties of spontaneous speech rarely conform to canonical patterns associated with specific phonetic segments, and (3) automatic-speech-recognition phonetic classifiers often require ca. 250 ms of acoustic context (spanning several segments) to function reliably. This pattern of evidence suggests that the essential cues for understanding spoken language are largely dynamic in nature, derived from (1) the complex modulation spectrum (incorporating both amplitude and phase) below 20 Hz, (2) segmentation of the speech signal into syllabic intervals between 50 and 400 ms, and (3) a multi-time-scale, coarse-grained analysis of phonetic constituents into features based on voicing, manner and place of articulation. [Work supported by the U.S. Department of Defense and NSF.]

4:20–5:00

#### Panel Discussion

TUESDAY AFTERNOON, 5 JUNE 2001

MONROE ROOM, 1:00 TO 4:35 P.M.

### Session 2pSP

#### Signal Processing in Acoustics and Underwater Acoustics: Bayesian Signal Processing Approach in Acoustics

James V. Candy, Cochair

*Lawrence Livermore National Laboratory, University of California, L-156, P.O. Box 808, Livermore, California 94551*

Ning Xiang, Cochair

*National Center for Physical Acoustics, University of Mississippi, P.O. Box 1848, University, Mississippi 38677*

Chair's Introduction—1:00

#### Invited Papers

1:05

**2pSP1. Bayesian signal processing in acoustics: detection, estimation and tracking.** Leon H. Sibul (Appl. Res. Lab., Penn State Univ., University Park, PA 16802, lhs2@psu.edu)

A tutorial introductory lecture on Bayesian signal processing for detection, estimation, and tracking is presented. After a brief historical overview, Bayes rule and risk are defined and used for development of detectors that minimize Bayes risk. Detectors that minimize Bayes risk are called Bayes detectors. Bayes risk in statistical signal processing is the expected cost of making a wrong decision. The decision process of deciding between mutually exclusive and exhaustive alternatives (i.e., an echo and noise are present versus noise only is present) is a binary hypothesis test or detector. We show how Bayes detectors are related to the maximum likelihood test and Neyman–Pearson and minimax criteria. Multiple hypotheses tests are also reviewed. Estimate of random parameters that minimize the risk are called Bayes estimates and the resulting risk, the Bayes risk. Minimum mean square error (MMSE), MAP (mode of the posteriori density) and other estimates can be derived using appropriate cost functions. Sequential application of Bayes rule can be used to derive Wiener and Kalman filters. Some of the basic difficulties and issues of Bayesian signal processing will be discussed. [Supported by ONR, Code 333, Les Jacobi, Program Officer.]

1:55

**2pSP2. Incorporating uncertainty in ocean acoustics for optimum signal detection.** Loren W. Nolte, Stacy L. Tantum, and Liewei Sha (Dept. of Elec. and Computer Eng., Duke Univ., Durham, NC 27708)

An overview of detection, classification and localization of signals in an uncertain ocean environment is presented from a Bayesian perspective. The Bayesian approach allows one to both model and incorporate, in an optimal way, the inherent uncertainty that often exists in the knowledge of ocean acoustic environmental parameters. The relationship between environmental parameter estimation and signal detection is illustrated, and it is shown how environmental parameter estimation can be viewed as an inherent part of the optimum Bayesian detector. Furthermore, by using the Bayesian approach, one can design optimum detection algorithms that are

robust with respect to precise knowledge of environmental parameters. Using the receiver operating characteristic (ROC), one can illustrate the trade-off among detection performance, environmental uncertainty, and signal-to-noise ratio. Several examples are taken from an uncertain shallow water environment to illustrate these trade-offs. Finally, it is shown how depth classification can be viewed as an optimal detection problem. [Work supported by ONR.]

2:15

**2pSP3. Towed array processing as a Bayesian problem.** Edmund J. Sullivan (OASIS, 5 Militia Dr., Lexington, MA 02421, sullivan@oasislex.com)

The towed array range and bearing estimation problem is cast in the form of a Bayesian estimation problem. It is shown that by casting the problem in the form of a MAP estimator, the Kalman estimator naturally follows. There are two advantages to this. First, the problem becomes recursive, resulting in an adaptive processor. Second, once the problem is in the Kalman form, it becomes possible to include sophisticated models of the signals and noise in a natural way. Thus, performance can be enhanced, since the models essentially provide *a priori* information to the processor. In this work an algorithm, which explicitly contains the forward motion of the array, has been developed. It is capable of performing bearing and range (wavefront curvature) estimation with a short array. Here, short means that the ratio of the physical aperture of the array to the range of interest is small. An example using simulated data is given where an array with a length of 45 meters is capable of determining the range and bearing of a narrow band source. Results are shown for several array speeds, signal-to-noise ratios bearings and ranges.

2:35–2:50 Break

2:50

**2pSP4. Bayesian inference in architectural acoustics: Estimation of decay times in coupled spaces.** Paul M. Goggans (Dept. of Elec. Eng., Univ. of Mississippi, University, MS 38677) and Ning Xiang (Univ. of Mississippi, University, MS 38677)

Sound energy in coupled spaces decays at multiple rates under certain conditions. Identifying decay times in these coupled spaces often demands considerable effort. Generally, different portions of multi-rate decay functions obtained from measured data cannot always be distinctly recognized. A model-based parameter estimation approach within the Bayesian framework facilitates the evaluation of multiple decay times through an extension of the decay model established in a previous work [N. Xiang, J. Acoust. Soc. Am. **98**, 2112–2121 (1995)]. Unlike gradient-based approaches, no careful estimation of initial values is required. Therefore, a robust algorithmic estimation of multiple decay times from experimentally measured decay functions shows advantages over the existing nonlinear regression approach.

3:10

**2pSP5. Separation and localization of acoustic signals.** Kevin H. Knuth (Ctr. for Adv. Brain Imaging, The Nathan Kline Inst., 140 Old Orangeburg Rd., Orangeburg, NY 10962)

The cocktail party problem is considered where there are several sound sources present in a free-field environment and an observer, who has the ability to record only linear mixtures of these sounds, is interested in obtaining estimates of the original source signals. Previous application of Bayesian inference to the general problem of blind source separation has demonstrated that it is possible to include additional prior information such as the transmission of the signals to the detectors and the possible locations of the sources to improve the source estimates. The derivation of prior probabilities describing such information will be discussed as will the construction of useful models and application of a general Bayesian methodology allowing one to deal with noninstantaneous inverse square transmission, noisy environments and any number of sources and detectors. Methods for dealing with environments in enclosure will also be described. [Work supported in part by NARSAD Young Investigator Award.]

3:30

**2pSP6. Bayesian estimation framework for source separation with mixture densities.** Ali Mohammad-Djafari and Hichem Snoussi (L2S, Supelec, Plateau de Moulon, 91192 Gif-sur-Yvette, France)

One of the main signal processing problems in acoustics applications is the sources separation. This problem is inherently an ill-posed problem. The Bayesian inference framework is a coherent way to solve such problems by modeling sources and canals and by combining prior information coming from these probabilistic modeling and information included in the data. In this contribution, after a brief presentation of general source separation problems and the Bayesian inference framework, we present new algorithms to source separation for the case of noisy instantaneous linear mixture, within the Bayesian estimation framework. The prior source distribution is modeled by a mixture of Gaussians and the mixing matrix elements distributions by a Gaussian. We model the mixture of Gaussians hierarchically by means of hidden variables representing the labels of the mixture. Then, we consider the joint *a posteriori* distribution of sources, mixing matrix elements, labels of the mixture, and other parameters of the mixture with appropriate prior probability laws to eliminate degeneracy of the likelihood function of variance parameters and we propose iterative algorithms to estimate jointly sources, mixing matrix, and hyperparameters: Joint MAP (maximum *a posteriori*) algorithms.

2p TUE. PM

3:50

**2pSP7. Acoustic inversion via linearization and Bayesian multipath identification.** Xiaoqun Ma and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102)

A linearization approach to acoustic inversion is proposed employing distinct ray paths (direct arrival, first surface bounce, and first bottom bounce) for source localization and bathymetry and sound speed estimation. The ray path arrivals are selected from broadband, shallow water, synthetic data using a Bayesian time delay estimation scheme calculating posterior probability density functions of the delays in an efficient way. A linear system is then formed relating unknown parameters and arrival time data. The regularization method is used for the solution of the linear system [S. E. Dosso *et al.*, J. Acoust. Soc. Am. **104**, 846–859 (1998)] with excellent results. Also a simple least-squares approach for the solution of the system is implemented; results of the two approaches are compared. Finally, the linearization multipath based technique is successfully applied to real acoustic broadband data for source and receiver localization, and bathymetry and sound speed estimation. [Work supported by ONR.]

4:05

**2pSP8. A Bayesian framework for perceptually motivated audio signal enhancement.** Patrick J. Wolfe and Simon J. Godsill (Signal Processing Group, Eng. Dept., Cambridge Univ., Trumpington St., Cambridge CB2 1PZ, UK, p.wolfe@ieee.org)

Spurred by the success of perceptual models in audio coding applications, researchers have begun to address audio signal enhancement in a similar manner. Here a statistical model-based approach is presented that uses cost functions based on auditory perception. As noise reduction inevitably occurs at the expense of signal resolution, why not take advantage

of human perception in order to optimize this trade-off? By mathematically incorporating the notion of perceived signal quality, Bayesian risk theory does just that. The Bayesian paradigm is shown to provide an ideal framework within which to formalize such an approach, as it is rigorous, powerful, and generalizable. At the same time it allows the incorporation not only of psychoacoustic optimality criteria, but also of additional non-perceptual prior information such as that concerning audio signal behavior in time and frequency. Importantly, it also permits sequential estimation for real-time noise reduction. Audio examples and the corresponding noise reduction software may be found at <http://www-sigproc.eng.cam.ac.uk/~pjw47>. [Material by the first author is based upon work supported by a U.S. National Science Foundation graduate research fellowship.]

4:20

**2pSP9. Acoustic benchmark analysis of GRASP optimum sonar search plans.** Donald R. DelBalso, Melvin A. Wagstaff, Mona A. Collins, James H. Leclerc, and Kevin P. Hempsteter (Naval Res. Lab, Stennis Space Center, MS 39529-5004)

An application of a genetic algorithm and Monte Carlo simulation with Bayesian detection statistics is used to optimize sonar search tracks in nonhomogeneous environments. The optimization metric is maximum cumulative detection probability for a specific sonar (passive or active) against a target with specified characteristics (acoustic and tactical) during a fixed time period. This new search planning capability is named GRASP for genetic range-dependent algorithm for search planning. The sensitivity of GRASP solutions to various ocean environments is examined under benchmark conditions, i.e., fairly simple synthetic environments and a simple target model. The results indicate that the genetic algorithm produces intuitive tracks that correlate highly with acoustic signal excess predictions, as expected.

TUESDAY AFTERNOON, 5 JUNE 2001

ADAMS ROOM, 1:25 TO 5:00 P.M.

## Session 2pUW

## Underwater Acoustics and Acoustical Oceanography: North Pacific Acoustic Laboratory Experiment

Peter F. Worcester, Chair

*Scripps Institute of Oceanography, University of California at San Diego, 9500 Gilman Drive,  
La Jolla, California 92093-0225*

Chair's Introduction—1:25

## Contributed Papers

1:30

**2pUW1. The North Pacific Acoustic Laboratory (NPAL) experiment.** The NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, P. F. Worcester) (SIO-UCSD, La Jolla, CA 92093; APL-UW, Seattle, WA 98105; and WHOI, Woods Hole, MA 02543, pworchester@ucsd.edu)

The North Pacific Acoustic Laboratory program augmented the existing ATOC acoustic network with a sparse, two-dimensional receiving array installed west of Sur Ridge, California, from July 1998 through June 1999, to receive transmissions from the 75-Hz ATOC source north of Kauai. The NPAL array consisted of four 20-element vertical arrays, each with a 700-m aperture, and one 40-element vertical array with a 1400-m

aperture. The arrays were deployed transverse to the 3900-km acoustic path from the Kauai source and had a total horizontal aperture of 3600 m. Data collected with the billboard array and the U.S. Navy SOSUS receivers are being used (i) to study the temporal, vertical, and horizontal coherence of long-range, low-frequency resolved rays and modes, (ii) to study 3-D propagation effects, (iii) to examine directional ambient noise properties, and (iv) to improve basin-scale ocean nowcasts via assimilation of acoustic data and other data types into models. In addition to acoustic data, environmental data along the path from the Kauai source to the billboard array were acquired by two oceanographic sub-surface moorings and by two XBT/CTD/ADCP transects along the path. The experiment will be described and some preliminary results presented.



**2pUW2. Extracting acoustic observables from the NPAL billboard array data.** Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA, mad@ucsd.edu) and the NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, P. F. Worcester) (SIO-UCSD, La Jolla, CA 92093; APL-UW, Seattle, WA 98105; and WHOI, Woods Hole, MA 02543)

Low-frequency (75-Hz) acoustic signals were repeatedly transmitted over a 1 year period and sampled vertically (with up to a 1400-m aperture) and horizontally (with a 3600-m cross-range aperture) by a distant billboard array (3900-km range) as described by the NPAL Group. The data are complicated by the fact that the sound interacts with the bottom near both the source and receiver. Vertical beamforming is used to filter the bottom interacting energy, and thus allow analysis of the fundamental acoustic properties. Subband and subarray processing is used to produce estimates of arrival times and angles or resolved ray arrivals. Time-series of acoustic travel-times and arrival angles are then developed.

## 2:00

**2pUW3. Analysis of mode coherence and intensity at megameter ranges.** Kathleen E. Wage (George Mason University, MS 1G5, 4400 University Dr., Fairfax, VA 22030) and the NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, P. F. Worcester)

The low-order modes constitute some of the most energetic arrivals at long ranges. Understanding fluctuations of these mode arrivals is crucial to their use as observables in matched field processing and tomography. Both simulated and experimental data indicate that at megameter ranges, the low modes have complex arrival patterns due to internal-wave-induced coupling. Analysis of broadband receptions at 3515 km from the ATOC experiment has shown that mode coherence times are on the order of 6 minutes and that centroid statistics provide useful measures of arrival time trends over the course of several months [Wage *et al.*, IEEE Sensor Array and Multichannel Signal Processing Workshop Proceedings, pp. 102–106, 2000]. The North Pacific Acoustic Laboratory (NPAL) experiment presents an opportunity for further research on broadband mode arrivals at megameter ranges. This study examines temporal coherence, intensity variations, and other mode statistics using data from the 40-element NPAL vertical line array. Experimental results are compared with PE simulations of propagation through internal waves of varying strengths, and the impact of the up-slope propagation near the receivers on the mode statistics is discussed.

## 2:15

**2pUW4. Observing horizontal wave fronts from the NPAL billboard array data.** Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA, mad@ucsd.edu) and the NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, P. F. Worcester) (SIO-UCSD, La Jolla, CA 92093; APL-UW, Seattle, WA 98105; and WHOI, Woods Hole, MA 02543)

One of the main objectives of the NPAL experiment is to investigate the horizontal refraction and coherence of the acoustic wave fronts at long range. Given time series of acoustic arrival times and angles of resolved ray arrival arrivals, a detailed look at the acoustic wave fronts is possible. First and second order statistics (density functions and coherences) of the wave fronts are investigated. The wave fronts are shown to vary with time, frequency, depth and across the horizontal aperture.

**2pUW5. Horizontal refraction of acoustic signals retrieved from NPAL billboard array data.** Alexander G. Voronovich, Vladimir E. Ostashev (NOAA/Environ. Technol. Lab., 325 Broadway, Boulder, CO 80305-3328), and the NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, P. F. Worcester)

A comprehensive, long-range sound propagation experiment was carried out with the use of the billboard acoustic array of the North Pacific Acoustic Laboratory (NPAL) in 1999. The antenna consisting of five vertical line arrays was deployed near a California coast and received broadband acoustic signals transmitted from Hawaii over a distance of about 4000 km. Acoustic signals propagating over such a long distance might exhibit noticeable horizontal refraction. The paper will present results of processing the NPAL data to infer horizontal refraction angle (HRA) as a function of time. Different methods were used for determining HRA. The first approach employed cross correlation of the acoustic signals at different VLAs. Time delay corresponding to maximum of cross correlation is related to HRA assuming the angle is approximately the same for all rays (or modes). The second method used modal representation of the arriving broadband signals. The dependency of the amplitudes of acoustic modes on mode number, frequency, and arrival angle was determined independently within narrow frequency bins, and then the results were averaged over whole frequency range. This method allowed, in particular, to evaluate angular width of the arrived signal, which appeared to be of the order of a few milliradians.

## 2:45

**2pUW6. The terminal problem: Model-data comparisons of basin scale transmissions including effects of shallow bathymetry near source and receiver.** Kevin D. Heaney (ORINCON Corp., 4350 N. Fairfax Dr., Arlington, VA 22203) and the NPAL Group (P. F. Worcester, J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel)

Acoustic transmissions on basin scale ranges are being used to determine depth-dependent temperature variability. With travel time being the primary observable, stationary sources and nearly stationary receivers are experimental requirements. This has led to the use of bottom-mounted sources and receivers to reduce travel time variability. The NPAL (North Pacific Acoustics Laboratory) experiment has transmitted broadband acoustic pulses from two bottom-mounted sources near the sound channel axis. Recordings have been taken on the NPAL billboard array, a linear series of five vertical line arrays moored in 1800 m of water near Monterey, CA. Additional recordings have been taken from the SOSUS system throughout the Pacific basin. The effects of the near source and near receiver scattering are examined. In particular, near source scattering leads to excess high-angle energy entering deep water with a travel time delay of nearly 1 s due to the low group speeds of high-angle rays/modes in shallow water. We also compare the energetics of the arriving rays that have bounced once on the rising seafloor near the NPAL receivers. Comparisons of models and data for bottom interacting acoustics lead us to the perennial issue of geoacoustic parameters.

## 3:00–3:15 Break

## 3:15

**2pUW7. On “ray-like” arrivals in the deep-ocean shadow zone of megameter range acoustic transmissions.** Brian Dushaw and the NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, Peter F. Worcester) (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105-6698)

Receptions of long-range acoustic transmissions by deep hydrophone arrays in the Pacific and Atlantic often have “ray-like” arrivals that occur in the shadow zone of the predicted time front. These “ray-like” arrivals can frequently be identified with the cusps of the predicted time front, but



the receivers are up to 750 m below the depth of the cusps. Preliminary calculations show that the observed acoustic energy is not accounted for by errors in the sound speed, leakage of acoustic energy below the cusps as predicted by the full wave equation, or scattering due to internal waves. Data obtained during experiments in the Atlantic and Pacific will be reviewed. Experiments that have been conducted with receivers of vertical line arrays have not had receivers deep enough to observe this phenomena. The effect is seen when bottom-mounted or midwater acoustic sources are used. These data present a number of problems: If the ray paths are wandering all over the water column, why are predictions of ray travel times usually accurate? How does the energy loss associated with these data increase the attenuation of very long-range acoustic transmissions? Without knowing the forward problem, how can these data be used to determine oceanographic information?

## 3:30

**2pUW8. Bottom grazing acoustic arrivals: Explanation of anomalous arriving deep multipaths.** Kevin D. Heaney and Henry Cox (Orincon Corp., 4350 N. Fairfax Dr., Arlington, VA 22203)

During the 1996 Acoustic Engineering Test (AET) of the ATOC project, receptions were recorded on deep SOSUS arrays in the central Pacific. With the given SNR, several pairs of doublet arrivals are visible. These doublets have the form one would expect from an axial source, but to a receiver much shallower in the water (~500–1000 m). The presence of this energy, well below the turning point of the corresponding rays has been termed the “acoustic shadowzone phenomena” [Dushaw *et al.*, IEEE J. Ocean. Eng. **24**, 202–214 (1999)]. Range independent normal mode modeling for an axial source reveals bottom grazing acoustic energy at all ranges. The arrival envelope (pairs of doublets) is independent of range. These arrivals are present in ray theory calculations when bottom grazing ray arrivals are included. With a critical angle of 18 deg, there are many subcritical bottom interacting rays with very little attenuation. This phenomenon has been understood in shallow-water acoustic problems where the reflection coefficient is linear with angle. The time spread of the pulse (number of rays) grows linear with range. The attenuation of the pulse decays linearly with range. The result is an envelope that is independent of range.

## 3:45

**2pUW9. Scattering into the shadow zone.** Walter Munk (Scripps Inst. of Oceanogr., Univ. of California–San Diego, La Jolla, CA 92093)

We consider the scattering of sound into the shadow zone in long-range transmission by processes other than internal waves. Microfrontal activity measured by Dan Rudnick in the north Pacific appears to be a significant factor.

## 4:00

**2pUW10. Does semiclassical ray theory work for megameter propagation?** Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

In the presence of internal waves, a very large number of rays connect any source to a distant receiver. The ray intensities vary by orders of magnitude, causing the scintillation index predicted by semiclassical ray theory to be very large. In this talk, possible diffractive effects at frequencies at or below 100 Hz are considered. One effect is caustics. A region where Airy functions must be used can be identified; if this region contains additional caustics, Airy functions do not suffice. This caustic region is identified and characterized from numerical ray-tracing simulations. The rays that are high intensity occur in narrow beams. The diffraction of these beams is estimated from the same numerical simulations. [Work supported by ONR.]

**2pUW11. Scintillations in long-range acoustic transmissions.** John Colosi (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Frederick Tappert (Rosenstiel School of Marine and Atmospheric Sci., Miami, FL 33149), and Matthew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA 92093)

In the Acoustic Thermometry of Ocean Climate (ATOC) program's Acoustic Engineering Test (AET), broadband 75-Hz center frequency transmissions were recorded on a 700-m-long vertical array, 3252 km distant. Previously reported results from the AET using 12.7-min. averaged data by Colosi *et al.* [J. Acoust. Soc. Am. **105**, 3202–3218 (1999), hereafter referred to as Colosi99] revealed surprisingly weak acoustic scattering for early arriving identifiable wave-fronts; these results have been confirmed using 1.8-min averaged data. It is shown that scintillation index (SI) is a strong function of position along the pulse with smallest values occurring at the peak and larger values occurring at the tails. Intensity PDFs of identifiable wave-fronts are reanalyzed in terms of both peak intensity and integrated pulse energy (IE), but both PDFs are very closely log-normal. Regarding multipathing along the wave-fronts it is found that on average there are 1.7 peaks per wave-front segment per hydrophone. The combined observation of weak scattering and multipathing is a novel result. Colosi99 analyzed the finale in terms of peak scintillations and found a near log-normal intensity PDF. Reprocessing the full field without limiting data to intensity peaks and accounting for mean intensity nonstationarity yields an exponential intensity PDF.

## 4:30

**2pUW12. Noise field statistics and coherence on the NPAL array.** Arthur B. Baggeroer (Dept. of Ocean Eng., MIT, Cambridge, MA 02139, abb@arctic.mit.edu), Edward K. Scheer (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and the NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, P. F. Worcester)

While the NPAL array was primarily deployed to examine the spatial coherence of the Hawaii source, it is also a rich data set for ambient noise studies. Shipping noise, earthquakes and biologics all have been identified in the NPAL data. Moreover, ambient noise coherence is the primary issue in maximizing the SNR output of a sonar system. The first and second order statistics of data from the NPAL “noise only” segments have been analyzed with the following results: (i) There is a wide spread in the peak levels, most likely due to the proximity to shipping lanes. The maximum peak level in the recording band is 117 dB. (ii) Full broadband coherences tend to be low because of the presence of many ships. (iii) If one examines frequency bands of 1–2 Hz, then lines of individual ships can be identified and associated and they are very coherent across NPAL aperture. (iv) Vertical beamforming indicates relatively highly directional spectra at low grazing angles and “noise notch” for the spectra at higher frequencies. Horizontal beamforming has been difficult to implement due to element positioning errors and the large array transit time.

## 4:45

**2pUW13. A comparison of ocean ambient sound levels after 30 years for a coastal site off California.** Rex K. Andrew, Bruce M. Howe, James A. Mercer (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98195-6698, randrew@apl.washington.edu), and Matthew A. Dzieciuch (Univ. of California at San Diego, La Jolla, CA 92093-0225), and the NPAL Group (J. A. Colosi, B. D. Cornuelle, B. D. Dushaw, M. A. Dzieciuch, B. M. Howe, J. A. Mercer, R. C. Spindel, P. F. Worcester)

As part of the North Pacific Acoustic Laboratory project, ambient sound data from 1994 to the present has been collected. Long-term averages of these data from a receiver on the continental slope west of Point Sur, CA, are compared to earlier measurements made at the same site over 1963–1965 by Wenz [Wenz, J. Underwater Acoust. **19** (1969)]. The levels

Wenz reported fall below our 10% quantile from 5 Hz to 50 Hz, rise to the 50% quantile (i.e., the median) at 100 Hz, and again fall below the 10% quantile by 250 Hz. Wenz removed highly variable “transient” data before calculating his averages. We mimicked his processing with the NPAL data and obtained a result which is virtually indistinguishable from the

median, which is approximately 1 dB below the (dB) mean of each one-third octave band. Hence, our median levels are directly comparable to Wenz’ results, and this comparison shows that the 1994–2000 levels exceed the 1963–1965 levels by 9 dB or less below 100 Hz and again at 250 Hz, but are roughly similar at 100 Hz. [Work sponsored by ONR.]

2p TUE. PM